JC14 Rec'd PCT/PTO 26 DEC 2001

FORM PTO-1390 (REV. 9-2001) U S DEPARTMENT OF COMMERCE PATENT AND TRADEMARK OFFICE **TRANSMITTAL LETTER TO THE UNITED STATES DESIGNATED/ELECTED OFFICE (DO/EO/US) CONCERNING A FILING UNDER 35 U.S.C. 371**		ATTORNEY 'S DOCKET NUMBER Mic.6055 U.S. APPLICATION NO (If known, see 37 CFR 1 5 10/030521
INTERNATIONAL APPLICATION NO. PCT/EP00/04798	INTERNATIONAL FILING DATE 26 May 2000	PRIORITY DATE CLAIMED 23 June 1999

TITLE OF INVENTION
A METHOD FOR PROCESSING AN AUDIO SIGNAL

APPLICANT(S) FOR DO/EO/US
Matthias VIERTHALER, Martin WINTERER

Applicant herewith submits to the United States Designated/Elected Office (DO/EO/US) the following items and other information:

1. [X] This is a **FIRST** submission of items concerning a filing under 35 U.S.C. 371.
2. [ ] This is a **SECOND** or **SUBSEQUENT** submission of items concerning a filing under 35 U.S.C. 371.
3. [X] This is an express request to begin national examination procedures (35 U.S.C. 371(f)). The submission must include items (5), (6), (9) and (21) indicated below.
4. [X] The US has been elected by the expiration of 19 months from the priority date (Article 31).
5. [X] A copy of the International Application as filed (35 U.S.C. 371(c)(2))
   a. [X] is attached hereto (required only if not communicated by the International Bureau).
   b. [ ] has been communicated by the International Bureau.
   c. [ ] is not required, as the application was filed in the United States Receiving Office (RO/US).
6. [X] An English language translation of the International Application as filed (35 U.S.C. 371(c)(2)).
   a. [X] is attached hereto.
   b. [ ] has been previously submitted under 35 U.S.C. 154(d)(4).
7. [ ] Amendments to the claims of the International Aplication under PCT Article 19 (35 U.S.C. 371(c)(3))
   a. [ ] are attached hereto (required only if not communicated by the International Bureau).
   b. [ ] have been communicated by the International Bureau.
   c. [ ] have not been made; however, the time limit for making such amendments has NOT expired.
   d. [ ] have not been made and will not be made.
8. [ ] An English language translation of the amendments to the claims under PCT Article 19 (35 U.S.C. 371 (c)(3)).
9. [ ] An oath or declaration of the inventor(s) (35 U.S.C. 371(c)(4)).
10. [X] An English lanugage translation of the annexes of the International Preliminary Examination Report under PCT Article 36 (35 U.S.C. 371(c)(5)).

Items 11 to 20 below concern document(s) or information included:

11. [X] An Information Disclosure Statement under 37 CFR 1.97 and 1.98.
12. [ ] An assignment document for recording. A separate cover sheet in compliance with 37 CFR 3.28 and 3.31 is included.
13. [X] A FIRST preliminary amendment.
14. [ ] A SECOND or SUBSEQUENT preliminary amendment.
15. [X] A substitute specification.
16. [ ] A change of power of attorney and/or address letter.
17. [ ] A computer-readable form of the sequence listing in accordance with PCT Rule 13ter.2 and 35 U.S.C. 1.821 - 1.825.
18. [ ] A second copy of the published international application under 35 U.S.C. 154(d)(4).
19. [ ] A second copy of the English language translation of the international application under 35 U.S.C. 154(d)(4).
20. [X] Other items or information:
   -International Preliminary Examination Report in German
   - Certificate of Express Mailing
   - Proposed Drawing Amendment

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U.S. APPLICATION NO. (if known, see 37 CFR 1.5)	INTERNATIONAL APPLICATION NO	ATTORNEY'S DOCKET NUMBER
10/030521	PCT/EP00/04798	Mic.6055

21. [X] The following fees are submitted:

	CALCULATIONS	PTO USE ONLY
BASIC NATIONAL FEE (37 CFR 1.492 (a) (1) - (5)):		
Neither international preliminary examination fee (37 CFR 1.482) nor international search fee (37 CFR 1.445(a)(2)) paid to USPTO and International Search Report not prepared by the EPO or JPO **$1040.00**		
International preliminary examination fee (37 CFR 1.482) not paid to USPTO but International Search Report prepared by the EPO or JPO **$890.00**		
International preliminary examination fee (37 CFR 1.482) not paid to USPTO but international search fee (37 CFR 1.445(a)(2)) paid to USPTO **$740.00**		
International preliminary examination fee (37 CFR 1.482) paid to USPTO but all claims did not satisfy provisions of PCT Article 33(1)-(4) **$710.00**		
International preliminary examination fee (37 CFR 1.482) paid to USPTO and all claims satisfied provisions of PCT Article 33(1)-(4) **$100.00**		
ENTER APPROPRIATE BASIC FEE AMOUNT =	$ 890.00	
Surcharge of **$130.00** for furnishing the oath or declaration later than [ ] 20 [X] 30 months from the earliest claimed priority date (37 CFR 1.492(e)).	$ 130.00	

CLAIMS	NUMBER FILED	NUMBER EXTRA	RATE	$	
Total claims	20 - 20 =	0	x **$18.00**	$	
Independent claims	4 - 3 =	1	x **$84.00**	$ 84.00	
MULTIPLE DEPENDENT CLAIM(S) (if applicable)			+ **$280.00**	$	
TOTAL OF ABOVE CALCULATIONS =				$1,104.00	
[ ] Applicant claims small entity status. See 37 CFR 1.27. The fees indicated above are reduced by 1/2. +				$	
SUBTOTAL =				$1,104.00	
Processing fee of **$130.00** for furnishing the English translation later than [ ] 20 [ ] 30 months from the earliest claimed priority date (37 CFR 1.492(f)).				$	
TOTAL NATIONAL FEE =				$1,104.00	
Fee for recording the enclosed assignment (37 CFR 1.21(h)). The assignment must be accompanied by an appropriate cover sheet (37 CFR 3.28, 3.31). **$40.00** per property +				$	
TOTAL FEES ENCLOSED =				$1,104.00	
				Amount to be refunded:	$
				charged:	$

a. [X] A check in the amount of $ 1,104.00 to cover the above fees is enclosed.

b. [ ] Please charge my Deposit Account No. ____________ in the amount of $ __________ to cover the above fees. A duplicate copy of this sheet is enclosed.

c. [X] The Commissioner is hereby authorized to charge any additional fees which may be required, or credit any overpayment to Deposit Account No. 19-0079. A duplicate copy of this sheet is enclosed.

d. [ ] Fees are to be charged to a credit card. **WARNING:** Information on this form may become public. **Credit card information should not be included on this form.** Provide credit card information and authorization on PTO-2038.

NOTE: Where an appropriate time limit under 37 CFR 1.494 or 1.495 has not been met, a petition to revive (37 CFR 1.137 (a) or (b)) must be filed and granted to restore the application to pending status.

SEND ALL CORRESPONDENCE TO:

Patrick J. O'Shea
Samuels, Gauthier & Stevens, LLP
225 Franklin Street, Suite 3300
Boston, MA 02110
Telephone: (617) 426-9180 ext. 121
Facsimile: (617) 426-2275

Patrick O'Shea
SIGNATURE

Patrick J. O'Shea
NAME

35,305
REGISTRATION NUMBER

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

APPLICANT: Vierthaler et al.

GROUP: Not yet assigned

INTERNATIONAL APPLN. NO.: PCT/EP00/04798

EXAMINER: Not yet assigned

SERIAL NO: Not yet assigned

INTERNATIONAL FILING DATE: 26 May 2000

FOR: METHOD FOR PROCESSING AN AUDIO SIGNAL

FIRST PRELIMINARY AMENDMENT

Entry of this preliminary amendment is respectfully requested to amend the claims and to specification prior to examination.

Table of Contents:

Preliminary to calculation of the filing fee, please amend the above-identified application as follows:

10/030521
JC13 Rec'd PCT/PTO 26 DEC 2001

Docket No.: Mic.6055

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

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INTERNATIONAL FILING DATE: 26 May 2000

FOR: A METHOD FOR PROCESSING AN AUDIO SIGNAL

PROPOSED DRAWING AMENDMENT

The applicant proposes drawing amendments to FIGs. 1-4 as shown in the attached redlined copy of FIGs. 1-4.

If a telephone interview could assist in the prosecution of this application, please call the undersigned attorney.

Respectfully submitted,

Patrick O'Shea

Patrick J. O'Shea
Registration No. 35,305
Samuels, Gauthier & Stevens LLP
225 Franklin Street, Suite 3300
Boston, MA 02110
Telephone: (617) 426-9180
Extension 121

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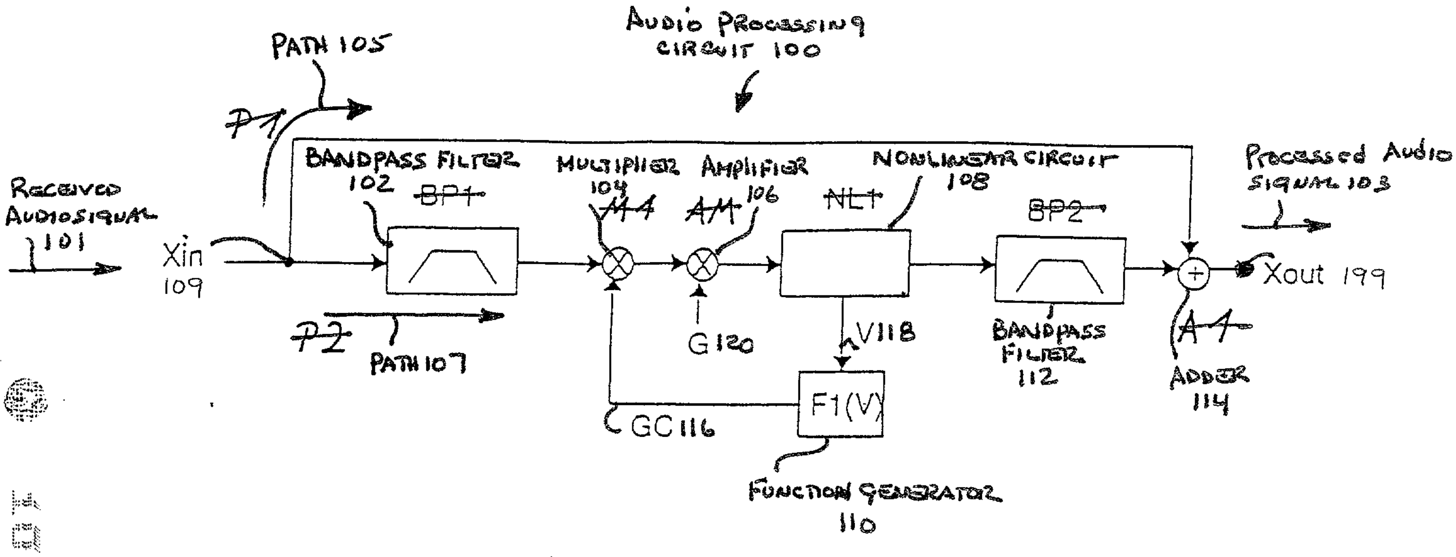

Fig. 1

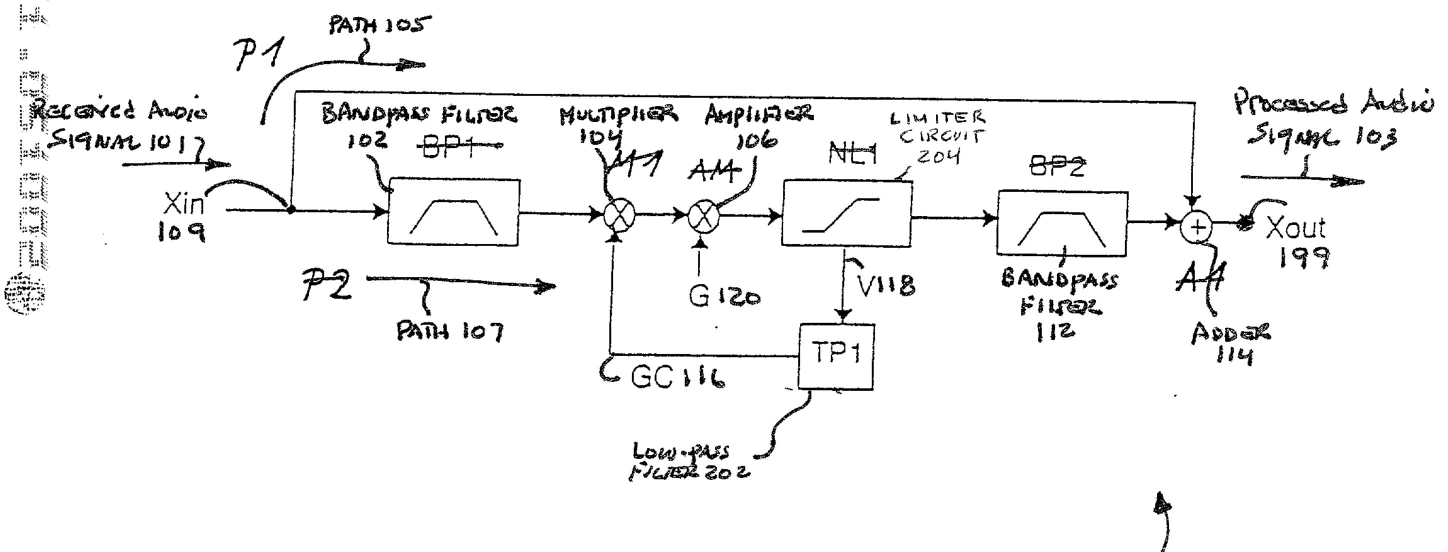

Fig. 2

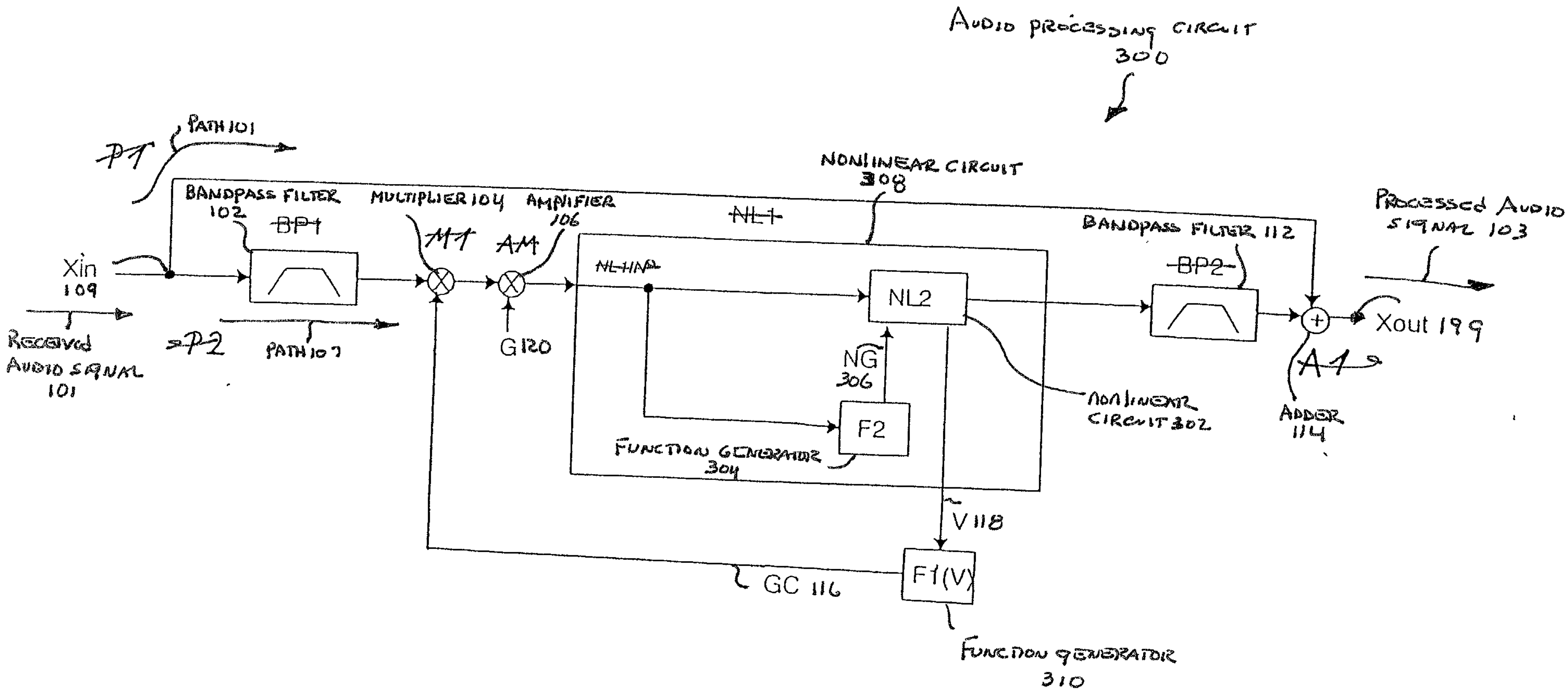

Fig. 3

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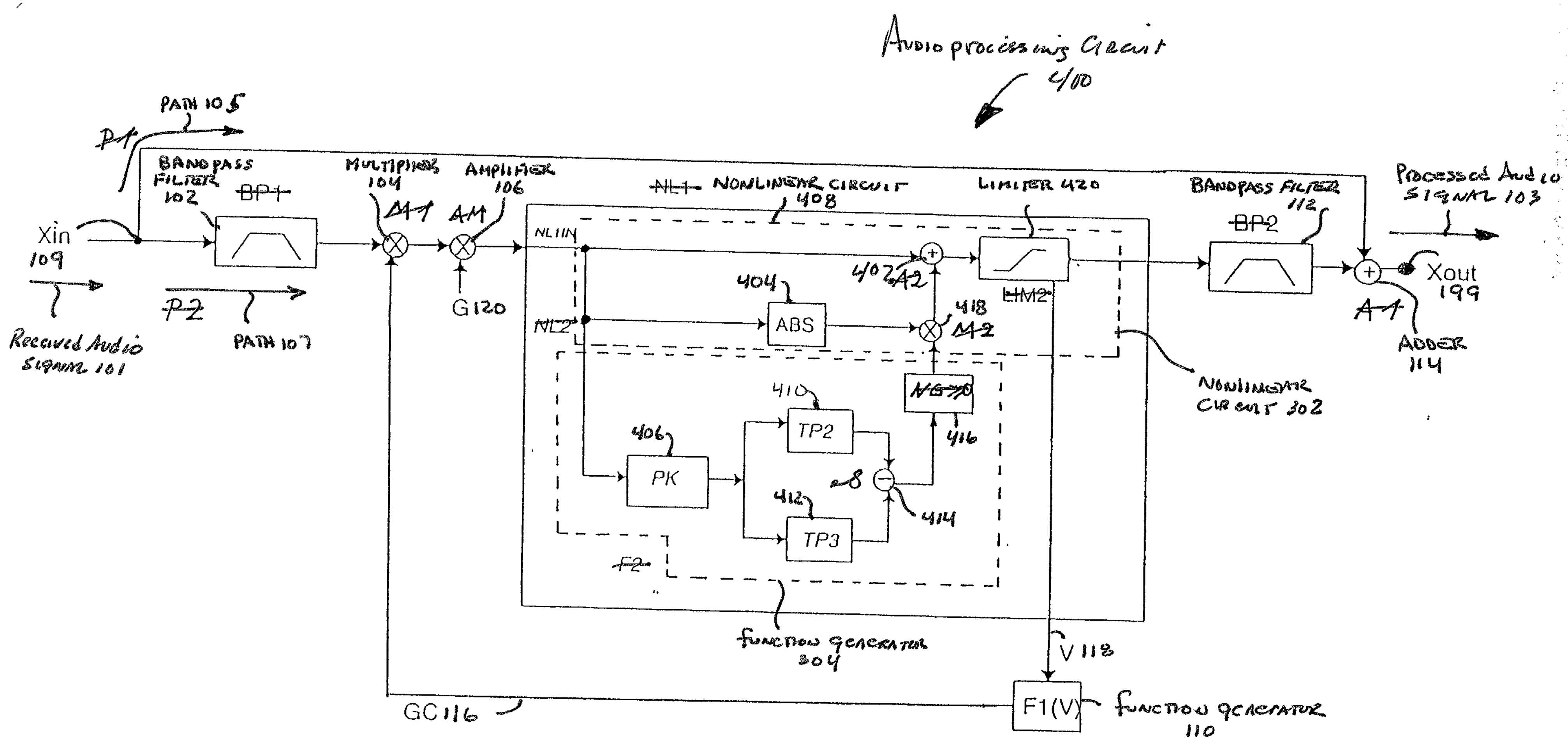

Audio processing Circuit 400
Xin 109
Received Audio signal 101
PATH 105
~~P1~~
Bandpass Filter 102
~~BP1~~
~~P2~~
PATH 107
Multiplier 104
~~M1~~
AM
Amplifier 106
G120
~~NL1~~ Nonlinear circuit 408
NL1IN
~~NL2~~
402
A2
404
ABS
418
~~M2~~
406
PK
410
TP2
412
TP3
414
~~NG~~
416
~~F2~~
Function generator 304
Limiter 420
~~LIM2~~
Nonlinear circuit 302
V 118
F1(V)
Function generator 110
GC116
Bandpass Filter 112
~~BP2~~
~~A1~~
Adder 114
Xout 199
Processed Audio signal 103

Fig. 4

Marked-up copy of the specification

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~~Description~~

~~A Method for Processing an Audio Signal~~

APPARATUS AND METHOD FOR PROCESSING AN AUDIO SIGNAL TO COMPENSATE FOR THE FREQUENCY RESPONSE OF LOUDSPEAKERS

BACKGROUND OF THE INVENTION

The present invention relates generally to processing audio signals and, more particularly, to a method and ~~to a circuit~~ apparatus for processing an audio signal to compensate for the frequency response of loudspeakers. ~~which is applied, via a first path, to the first input of an adder.~~

Devices that reproduce ~~Such methods and circuits are used in devices for~~ acoustic ~~reproduction~~signals~~,~~ such as, for example, ~~e.g.~~ television sets, radio receivers, ~~or~~ and stereo systems, include a circuit for processing the audio signals. Typically, such circuits are designed to compensate for the frequency response of the loudspeakers to improve acoustic reproduction, and to prevent overdriving the device or the system.

The most critical element in a unit for acoustic reproduction is the loudspeaker. The ~~, whose~~ acoustic pressure of a loudspeaker drops about 40 db per decade below a structurally determined limit frequency. This corresponds to the transfer function of a second-order filter. On the other hand, bass reflex and transmission line loudspeakers have transfer functions which correspond to a higher order filter. Their lower limit frequency typically lies between about 50 Hz and 200 Hz. The lower the limit frequency of a loudspeaker, the more expensive it is ~~it~~ to produce. Economical units~~,~~ such as, for example, ~~e.g.~~ television or portable radio receivers consequently are equipped with simple~~r~~ loudspeakers that have a relatively high ~~, whose~~ lower limit frequency ~~is relatively high~~. To improve acoustic reproduction in the lower frequency range,

the limit frequency of such units is shifted downward by preamplifying the low frequencies. However, this can cause the final amplifier and the loudspeakers to be overdriven. To prevent overdriving and possibly even destruction of the final amplifier or the loudspeaker, the output signal of the bass amplifier is fed back in such a way that the amplification of the lower frequencies is reduced if the output signal is strong. One example of such an approach is disclosed in ~~Such a method is known from the~~ U.S. ~~PS~~ Patent 5,305,388.

Another conventional approach is disclosed in ~~The~~ U.S. ~~PS~~ Patent 5,359,665. This patent describes a circuit in which the audio signal is conducted directly, via a first path, to the first input of an adder. The audio signal is simultaneously ~~, while at the same time it is~~ conducted, via a second path ~~, and via a low-pass filter and an amplifier with variable amplification,~~ to the second input of the adder. This second path includes a low-pass filter and an amplifier with variable amplification. The output of the amplifier is fed back, through a signal level detector, to its control input. This ~~procedure~~ circuit arrangement is described in the '665 patent to have the benefit of reducing ~~reduces~~ overdrive of the final amplifier.

From research results of psychoacoustics, it is known that a person can still unambiguously determine the fundamental ~~tone~~ level of a tone even when only harmonics of the fundamental frequency and not ~~if~~ the fundamental frequency itself is ~~not even~~ present in the spectrum~~, but only harmonics of the fundamental~~. This psychoacoustic effect is utilized in that the harmonic of the fundamental frequency is generated and is conducted to a loudspeaker whose limit frequency lies above this fundamental frequency. A listener consequently believes that s/he hears this low fundamental frequency even though the loudspeaker does not radiate it at all.

~~The~~ U.S. Patent ~~PS~~ 5,668,885 describes a circuit which thereby "entices" from a loudspeaker ~~with~~ having a relatively high lower limit frequency ~~still lower~~ frequencies which are

lower than its limit frequency. This is done by generating harmonics of the lower frequencies. The audio signal is conducted, via a first path, to the first input of an adder. In a second path, the audio signal passes through a low-pass filter, is rectified, ~~once again~~ passes through a second low-pass filter, is amplified, and finally is conducted to the second input of the adder.

~~The~~ U.S. Patent ~~PS~~ 4,150,253 likewise describes a method and a circuit, in which an audio signal is divided into two signal paths. In the first path, the audio signal traverses a high-pass filter, so as to shift the phase based on ~~in dependence on~~ the frequency. Those signals at the output of the high-pass filter~~,~~ ~~whose~~ which are at levels that exceed a given value~~,~~ are conducted to the input of a generator ~~to~~ which generates the harmonics of the fundamental frequency. The level of the signals at the output of the generator is attenuated to a value below the level of the original audio signal. This attenuated signal and the original audio signal are then added together.

~~The~~ U.S. Patent ~~PS~~ 4,700,390 describes a so-called synthesizer, in which harmonics are generated both for the lower and higher frequencies, and are added to the original audio signal. The alleged benefit set out in the '390 patent is that this ~~This is supposed to~~ achieves better reproduction both in the low and in the high frequency range.

~~The~~ U.S. ~~PS~~ Patent 5,771,296 likewise describes a circuit in which the audio signal is conducted, via a first path, directly into an adder, while, via a second path, the harmonics of the lower frequencies are generated and are added in the adder to the original signal. This provides the perception to the ~~, so as to make the~~ listener ~~believe~~ that ~~a~~ the loudspeaker radiates lower frequencies than it actually does.

Finally, ~~the~~ U.S. ~~PS~~ Patent 4,739,514 describes another circuit to improve the acoustic reproduction of low frequencies. With this circuit, too, the audio signal is conducted, via a first path, to the first input of an adder, while, via a second path consisting of an amplifier with

variable amplification in series with a bandpass filter, it is conducted to the second input of the adder. A signal level detector, whose input receives the audio signal, controls the amplification of the amplifier. The above and other conventional ~~referenced, known~~ methods and circuits have the disadvantage that, due to feedback technique employed, they react relatively slowly to rising amplitudes and, despite the feedback they tend to overdrive the device or the system. What is needed, therefore,

~~It is therefore the object of the invention to design a method in accordance with the generic part of Claim 1 and a circuit in accordance with the generic part of Claim 9, in such a fashion that the~~ is an audio processing technique that compensates for the frequency response of a loudspeaker ~~is compensated, its acoustic reproduction is improved, and~~ without overdriving ~~overdrive of~~ the circuit, its components or entire reproduction system ~~is prevented~~, ~~especially~~ particularly in the range of low frequencies.

SUMMARY OF THE INVENTION

Briefly, according to an aspect of the invention, a method for processing a received audio signal is disclosed. The method includes band-limiting the received audio signal to generate a first intermediate signal; multiplying the first intermediate signal by a correction factor to generate a second intermediate signal; amplifying the second intermediate signal by an amplification factor to generate a third intermediate signal; limiting the amplitude of the third intermediate signal to a specified maximum value to generate a fourth intermediate signal; band-limiting the fourth intermediate signal to generate a fifth intermediate signal; and adding the fifth intermediate signal to the received audio signal.

In another aspect of the invention, a circuit for processing an input audio signal received at an input of the circuit to provide at an output of the circuit a processed audio signal is disclosed. The circuit includes a first adder having first and second inputs and an output at which the processed audio signal is provided; a first conductive path connecting the circuit input to the first input of the first adder, the first conductive path constructed and arranged to deliver the received audio signal unaltered to the first adder; and a second conductive path connecting the circuit input to the second input of the first adder. The second conductive path includes a first bandpass filter having an output and an input connected to the circuit input; a multiplier having a first input connected to the first bandpass filter output, and a second input. and an output; a variable amplifier, having an output and an input connected to the multiplier output, for amplifying a signal received at the amplifier input in accordance with an amplification factor presented at a control input of the amplifier: a first nonlinear circuit having an output and an input connected to the amplifier output, the nonlinear circuit limiting to a specified maximum the amplitude of a signal presented at the first nonlinear circuit input; a second bandpass filter having an input connected to the nonlinear circuit output and an output defining the circuit network output; and a first function generator having an input connected to a control output of the first nonlinear circuit, and an output connected to the multiplier second input.

In a further aspect of the invention, a circuit for processing an input audio signal received at an input of the circuit to provide at an output of the circuit a processed audio signal. The circuit band-limits the received audio signal to generate a first intermediate signal; multiplies the first intermediate signal by a correction factor to generate a second intermediate signal; amplifies the second intermediate signal by an amplification factor to generate a third intermediate signal; limits the amplitude of the third intermediate signal to a specified maximum value to generate a fourth

intermediate signal; band-limits the fourth intermediate signal to generate a fifth intermediate signal; and adds the fifth intermediate signal to the received audio signal.

In a still further aspect of the invention, a circuit for processing an input audio signal received at an input of the circuit to provide at an output of the circuit a processed audio signal, is disclosed. The circuit includes a first conductive path through which the received audio signal travels; and a second conductive path through which the received audio signal travels, wherein the audio signal is processed such that harmonics of the signal components with a low-frequency are generated in the second path and are admixed to the signal in the first path, wherein in the second path the audio signal is sequentially bandpass filtered, weighted with a correction factor, amplified, limited to a maximum value, and bandpass filtered, wherein the correction factor is reduced when the maximum value is exceeded.

These and other objects, features and advantages of the present invention will become more apparent in light of the following detailed description of preferred embodiments thereof, as illustrated in the accompanying drawings.

~~The invention achieves this object in terms of method in that the audio signal is band-limited in a second path by means of a first bandpass filter , and that, at the output of the first bandpass filter, the band-limited audio signal is multiplied by a correction factor, is amplified by an amplification factor, and then its amplitude is limited to a specified maximum by means of a nonlinear circuit, and that the correction factor is reduced when the prescribed maximum is exceeded, but otherwise remains constant or is increased, and that the audio signal at the output of the first nonlinear circuit is band-limited by means of a second bandpass filter, and that the band-limited audio signal at the output of the second bandpass filter is added, in the adder, to the audio signal of the first path.~~

~~The invention achieves this object in terms of apparatus in that the audio signal is conducted to the second input of the first adder via a second path, consisting of a first bandpass filter, a first multiplier, an amplifier with variable amplification, a first nonlinear circuit, and a second bandpass filter, all connected in series, and that a control output of the first nonlinear circuit is connected to the input of a first function generator, whose output is connected to the first multiplier, and that an amplification is applied to the control input of the amplifier.~~

~~The invention will now be described and explained in terms of the inventive embodiments shown in the figures.~~

BRIEF DESCRIPTION OF THE DRAWING

FIG.~~igure~~ 1 is a schematic block diagram illustration of one ~~shows a first~~ embodiment of an audio processing circuit of the present ~~the~~ invention;~~.~~

FIG.~~igure~~ 2 is a schematic block diagram illustration of another ~~shows a second~~ embodiment of an audio processing circuit of the present ~~the~~ invention;~~.~~

FIG.~~igure~~ 3 is a schematic block diagram illustration of a further ~~shows a third~~ embodiment of an audio processing circuit of the present ~~the~~ invention~~.~~; and

FIG.~~igure~~ 4 is a schematic block diagram illustration of a still further ~~shows~~ a fourth embodiment of an audio processing circuit of the present ~~the~~ invention.

DETAILED DESCRIPTION OF THE INVENTION

The present invention is directed to an improved method and circuit for processing audio signals in a manner that compensates for the frequency response of the loudspeaker without overdriving the circuit components. Aspects of the present invention are described in detail below

~~inventive method will be described and explained~~ with reference to different ~~in terms of the first~~ embodiments of an audio processing circuit.

FIG. 1 is a schematic block diagram of one embodiment of au audio processing circuit of the present invention.~~the inventive circuit, shown in Figure 1.~~

In FIG.~~igure~~ 1, an ~~an~~ audio signal ~~Xin~~ 101 is received at circuit input Xin 109. Received audio signal 101 is conducted, via a first path 105~~P1~~, to a~~the~~ first input of an ~~a first~~ adder ~~A1~~ 114 and, via a second path 107~~P2~~, to a ~~the~~ second input of ~~the~~ adder 114~~A1~~. ~~The p~~Second path 107~~P2~~ includes ~~consists of~~ a first bandpass filter 102~~BP1~~, a ~~first~~ multiplier 104~~M1~~, an amplifier 106~~AM~~ with variable amplification, a ~~first~~ nonlinear circuit 108~~NL1~~, and a second bandpass filter 112,~~BP2~~ all connected in series. The processed ~~output~~ audio signal ~~Xout~~ 103 ~~can be taken from~~ is available at circuit output node Xout 199. The circuit output 199 is directly connected to the output of ~~the~~ adder 114~~A1~~. A control output of ~~the first~~ nonlinear circuit ~~NL1~~ 108 is connected to the input of a function generator ~~F1~~110. Nonlinear circuit 108 generates at its control output a control variable ("V") 118 which generates a correction factor ("GC") 116 based on the value of control variable 118. The output of ~~the~~ function generator ~~F1~~ 110 is connected to ~~the first~~ multiplier, 104~~M1.~~ providing correction factor ("GC") 116 to multiplier 104. As shown in Figure 1, a~~A~~ control variable ("G") 120 is applied to ~~the~~ a control input of ~~the~~ amplifier 106~~AM~~.

During operation, the ~~The~~ received audio signal 101~~Xin~~ is band-limited by ~~the first~~ bandpass filter ~~BP1~~102. ~~Then t~~The audio signal is then multiplied by ~~a~~ variable correction factor GC 116 in ~~the~~ multiplier 104~~M1~~. The resulting product provided at the output of ~~the~~ multiplier 104~~M1~~ is amplified in ~~the~~ amplifier ~~AM~~ 106 by ~~the~~ amplification factor G 120. ~~The n~~Nonlinear circuit ~~NL1~~ 108 limits the amplitude of the audio signal delivered by ~~the~~ amplifier ~~AM~~ 106 to a specified value. Embodiments of nonlinear circuit 108 are described in detail below. The output

signal of ~~the~~ nonlinear circuit ~~NL1~~ 108 is band-limited by ~~means of the~~ bandpass filter ~~BP2~~112. ~~The n~~As noted, nonlinear circuit ~~NL1~~ 108 ~~creates~~ generates ~~a~~ control variable V 118 the value of which is used by~~, from which the~~ function generator ~~F1~~ 110 to generate ~~generates the~~ correction factor GC 116. ~~This c~~Correction factor GC 116 is varied by ~~the~~ function generator ~~F1~~ 110 based on the value of ~~in dependence on the~~ control variable~~s V~~ 118. The value of correction factor GC 116 is ~~, in such a way that it is~~ reduced by function generator 110 when control variable 118 is greater than a predetermined maximum value, a condition that can result in an ~~in the case of~~ overdrive condition. On the other hand, if the level of the audio signal lies within ~~its~~ allowed limits, ~~the~~ correction factor GC 116 is increased by ~~the~~ function generator 110.~~F1,~~ Preferably, the value of correction factor GC 116 is limited to a maximum value of one (1).~~but at most up to the value 1.~~ This is described in greater detail below.

A schematic block diagram of another ~~The second~~ embodiment of the audio processing circuit of the present invention, referred to as audio processing circuit 200, is depicted ~~shown~~ in FIG.~~igure~~ 2. Audio processing circuit 200~~,~~ will now be described ~~and will be explained~~ in conjunction with the ~~first~~ embodiment of audio processing circuit 100 described above and illustrated in ~~of~~ FIG.~~igure~~ 1.

As shown in FIG. 2, the arrangement of the components of audio processing circuit 200 is similar to that of audio processing circuit 100. However, in the exemplary~~In the second~~ embodiment illustrated in Figure 2, ~~the~~ function generator 110~~F1~~ is implemented as a low-pass filter~~TP1~~. This function generator is referred to in FIG. 2 as low-pass filter 210. Also, in this embodiment, ~~and the first~~ nonlinear circuit ~~NL1~~ 108 is implemented as a limiter circuit~~,~~ which cuts off the signal amplitude above a specified threshold. This embodiment of nonlinear circuit 108 is referred to in FIG. 2 as limiter circuit 204. It should be understood to those of ordinary

skill in the art that the above-noted components of audio processing circuit 100 and audio processing circuit 200 can be implemented with any well-known circuit components now or later developed.

If the amplitude of the audio signal amplified by amplifier 106 ~~signal amplitude~~ exceeds the specified threshold, ~~the~~ nonlinear circuit ~~NL1~~ 108, 204 sets the value of ~~conducts the~~ control variable V 118 to a ~~with~~ negative value ~~V1 to the low-pass filter TP1~~. On the other hand, when the amplitude of ~~If~~ the signal ~~amplitude~~ lies below the specified threshold, ~~the~~ nonlinear circuit ~~NL1~~ 108, 204 generates a control variable V 118 ~~with~~ having a positive value ~~V2~~. As noted, control variable V 118 is received by low-pass filter 210 which generates correction factor 116 based on the value of control variable 118. ~~The c~~In accordance with the embodiment illustrated in FIG. 2, correction factor GC 116 for ~~the~~ multiplier ~~M1~~ 104 is created by filtering ~~the~~ control variable V 118 by ~~means of the~~ low-pass filter 210~~TP1~~.

The nonlinear operation in ~~the~~ nonlinear circuit 108~~NL1~~, which limits the amplitudes of the audio signal to a specified threshold, generates audio signals with ~~the~~ lower frequencies, which are also called harmonics of the bass signal. The shape of these harmonics is determined by the selection of ~~the choice of~~ the nonlinear operation implemented in ~~the~~ nonlinear circuit 108~~NL1~~ and by the dimensioning of ~~the~~ bandpass filter 112~~BP2~~. A useful form of these harmonics can be determined, for example, ~~e.g.~~ by calculation or by experiment, so as to make the beginning of an audio signal with low frequencies, ~~e.g.~~ such as the striking of a drum, appear clearer and brighter to a listener. The choice of function implemented in ~~of the~~ function generator 110, 210 ~~F1~~ determines the time which passes between the beginning of a strong, low-frequency tone and the reduction of ~~the~~ correction factor GC 116 to such an extent that ~~the~~ nonlinear circuit ~~NL1~~ 108 no longer generates harmonics. The length of this time interval, which is regarded as a time constant,

is determined by the dimensioning of ~~the~~ low-pass filter 210~~TP1~~ and the choice of the positive and negative values of ~~two~~ control variable Vs ~~V1 and V2~~ 118.

One advantage of audio processing circuits 100, 200 is that, ~~The inventive methods described above achieve the following effects:~~

with ~~With~~ a small signal amplitude, ~~the~~ amplifier 106 operates at full amplification and thus partially compensates the frequency characteristic of a loudspeaker. On the other hand, if the signal amplitude is sufficiently large, the frequency characteristic of the loudspeaker can be slightly compensated ~~only slightly, because otherwise it would be~~ to avoid the loudspeaker from being overdriven. Thus, upon ~~Upon~~ the onset of a bass signal, the~~is~~ bass signal is enriched with harmonics~~,~~ so that a listener, despite the lack of bass volume from the loudspeaker, has the sensation of clearly and loudly hearing the bass frequencies.

~~The third inventive embodiment, shown in Figure 3, will now be described and explained.~~

A further embodiment of the audio processing circuit of the present invention, referred to as audio processing circuit 300, is shown in FIG. 3. Audio processing circuit 300 will now be described in conjunction with the embodiments of audio processing circuits 100 and 200 described above and illustrated in FIGs. 1 and 2, respectively.

In the exemplary embodiment illustrated ~~the third embodiment,~~ in FIG.~~igure~~ 3, ~~the~~ a more detailed illustration of one embodiment of nonlinear circuit 108~~NL1~~ is shown ~~in detail~~. ~~The n~~This embodiment of nonlinear circuit 108~~NL1~~ is composed of a nonlinear circuit 302~~NL2~~ and a function generator 304~~F2~~. The ~~input~~ audio signal provided to the input of ~~the~~ nonlinear circuit ~~NL1~~ 108 is, as noted, ~~that is~~ the output signal of ~~the~~ amplifier 106. This signal ~~AM~~ is directly conducted to the input of ~~the~~ nonlinear circuit 302~~NL2~~ and to the input of ~~the~~ function generator 304. ~~F2,~~ The output of function generator 304 ~~whose output~~ is connected to ~~the~~ a control input of

~~the~~ nonlinear circuit 302~~NL2~~. The signal output of ~~the~~ nonlinear circuit 302~~NL2~~ is connected to the input of ~~the~~ bandpass filter 112~~BP2~~, while ~~the~~ a control output of ~~the~~ nonlinear circuit 302 ~~NL2~~ is connected to a ~~the~~ function generator 310. Function generator 310 can be implemented as function generator 110 described above with reference to FIG. 1, or as ~~F1 or to the~~ low-pass filter 210~~TP1~~ described above with reference to FIG. 2.

~~The n~~Nonlinear circuit ~~NL2~~ 302 continuously generates harmonics of the low-frequency components of the audio signal, which are weighted with ~~the~~ variable factor ("NG") 306 by ~~the~~ function generator 304~~F2~~. The factor NG 306 is a function of the signal input ~~signal~~to function generator 304. Depending on the choice of function for ~~the~~ function generator 304~~F2~~, manifold acoustic effects can be created.

For example, ~~the~~ function generator 304~~F2~~ can be designed so that ~~the~~ nonlinear circuit 302~~NL2~~ more strongly generates harmonics as soon as the signal amplitude ~~must~~ is to be limited, so as to prevent overdrive. In this way, ~~the~~ signal energy is distributed among the higher frequency harmonics, which a loudspeaker or a loudspeaker system can more accurately reproduce ~~better~~. Although ~~now~~ the energy of the lower frequency signal components is reduced in this approach, the listener nevertheless has the impression of a full bass sound due to the above-mentioned psychoacoustic effects.

~~The fourth inventive embodiment, shown in Figure 4, will now be described and explained.~~

~~Figure 4 shows in detail an exemplary structure of the nonlinear circuit NL2 and an exemplary structure of the function generator F2.~~

A still further embodiment of the audio processing circuit of the present invention, referred to as audio processing circuit 400, is shown in FIG. 4. Audio processing circuit 400 will now be described in conjunction with the embodiments of audio processing circuit 100, 200 and 300

described above and illustrated in FIGs. 1, 2 and 3, respectively. In this exemplary illustration of audio processing circuit 400, one embodiment of a detailed implementation of nonlinear circuit 308 is illustrated. This illustrative embodiment is referred to in FIG. 4 as nonlinear circuit 408. Nonlinear circuit 408 includes embodiments of nonlinear circuit 302, referred to as nonlinear circuit 402 herein, and function generator 304, referred to herein as function generator 404.

In this embodiment, the ~~The input~~ signal input to ~~the~~ nonlinear circuit 408~~NL1,~~ which, as noted, is the output signal from ~~the~~ amplifier ~~AM~~106, is conducted to ~~the~~ a first input of an adder 402~~A2~~, to the input of an absolute value forming circuit ~~ABS~~404, and to the input of a peak value detector ~~PK~~406.~~,~~ ~~whose~~ The output of peak value detector 406 is connected to the input of a low-pass filter 410~~TP2~~ and a low-pass filter 412~~TP3~~. The output of ~~the~~ low-pass filter 410~~TP2~~ is connected to a~~the~~ first input of a subtractor ~~S~~414, and the output of ~~the~~ low-pass filter 412~~TP3~~ is connected to a~~the~~ second input of ~~a~~ subtractor 414~~S~~. The output of ~~the~~ subtractor 414~~S~~ is connected, via a limiter 416,~~LIM1,~~ to the first input of a multiplier 418~~M2~~.

The output of ~~the~~ absolute value forming circuit ~~ABS~~ 404 is connected to the second input of ~~the~~ multiplier 418~~M2~~, the ~~whose~~ output of which is connected to a ~~the~~ second input of ~~the~~ adder 402~~A2~~. The output of ~~the~~ adder 402~~A2~~ is connected to the input of a limiter ~~LIM2~~420.~~,~~ The ~~whose~~ control output of limiter 420 is provided ~~delivers the control variable V~~ to ~~the~~ function generator 410. As with function generator 310, function generator 410 can be implemented as function generator 110 or as ~~F1 or to the~~ low-pass filter 210.~~TP1,~~ The ~~and whose~~ output of limiter 420 is connected to the input of ~~the~~ bandpass filter 112~~BP2~~. The processed audio signal 103 for a loudspeaker or a loudspeaker system is available at the output of ~~the~~ bandpass filter 112~~BP2~~.

~~The p~~Peak value detector ~~PK~~ 406 determines the level of the maximum amplitude occurring

during a specified time interval T. The output signal of ~~the~~ peak value detector ~~PK~~ 406 is time-averaged by the two low-pass filters 410 and 412. ~~TP2 and TP3.~~ In one embodiment, the ~~The~~ time constant of ~~the~~ low-pass filter ~~TP3~~ 412 is smaller than that of ~~the~~ low-pass filter 410. ~~TP2, i.e.~~ That is, the cut-off frequency of ~~the~~ low-pass filter 412 ~~TP3~~ with the smaller time constant is higher than that of ~~the~~ low-pass filter ~~TP2~~ 410 with the larger time constant. Because of the smaller time constant, the output signal of ~~the~~ low-pass filter ~~TP3~~ 412 follows a change of the input signal faster than does the output signal of ~~the~~ low-pass filter ~~TP2~~412. The absolute value forming circuit ~~ABS~~ 404 forms the absolute value of the input signal, which is weighted in ~~the~~ multiplier 418 ~~M2~~ by ~~the~~ factor NG 306~~,~~ ~~which has been~~ generated by ~~the~~ subtractor 416~~S~~. Limiter 416 ~~A limiter LIM1~~ limits ~~the~~ factor NG 306 to a ~~ar~~range between 0 and 1. The weighted absolute value of the input signal is added in ~~the~~ adder ~~A2~~ 402 to the input signal, and the resulting sum is limited to a specified amplitude by ~~means of the~~ limiter ~~LIM2~~420~~,~~ so as to prevent overdrive.

For example, if the amplitude of the input signal rises discontinuously, the level at the output of ~~the~~ low-pass filter ~~TP3~~ 412 will rise faster, due to its smaller time constant, than at the output of ~~the~~ low-pass filter ~~TP2~~410. As a result, ~~the~~ factor NG 306, which is to be regarded as a control variable, assumes a positive value for rising amplitudes in the input signal. As the rate at which the ~~The stronger the~~ amplitude of the input signal rises increases, ~~the~~ more harmonics will be generated and will be admixed to the input signal. On the other hand, if the amplitude falls, ~~the~~ factor NG 306 becomes negative, because now the level at the output of ~~the~~ low-pass filter 412~~TP3~~, due to its smaller time constant, becomes smaller than the level at the output of ~~the~~ low-pass filter 410~~TP2~~. Because the factor NG 306 has a lower limit of zero, no harmonics are admixed to the audio signal when the amplitudes are falling.

A significant advantage of the invention is that the nonlinear operation of ~~the~~ nonlinear circuit 408~~NL1~~, and the function of ~~the~~ function generator 410 ~~F1~~, determine the form of the harmonics as well as the time of their generation. It should be understood by those of ordinary skill in the art that adjustments in ~~By skillful choice of~~ the nonlinear operation of the nonlinear circuit 108, 208, 308, 408 and of the function of ~~the~~ function generator 110, 210, 310, 410, the invention can easily be adapted to loudspeakers with different characteristics, so that optimum compensation of the frequency response of a loudspeaker is always achieved. Because the amplitude of the audio signal is limited to a specified value by ~~the~~ nonlinear circuit ~~NL1~~, the inventive circuit reacts much faster than the prior art to rising amplitudes of the audio signal.

The invention is especially suited for acoustic reproduction units, e.g. television units, portable radios, which are equipped with loudspeakers with a weak bass range, because the invention prevents overdriving the entire reproduction system and at the same time offers the listener the illusion of sonorous basses, even though the loudspeakers really do not radiate these low bass frequencies.

Although the present invention has been shown and described with respect to several preferred embodiments thereof, various changes, omissions and additions to the form and detail thereof, may be made therein, without departing from the spirit and scope of the invention.

What is claimed is:

~~List of Reference Symbols~~

~~P1 First path~~
~~P2 Second path~~
~~BP1 First bandpass filter~~
~~BP2 Second bandpass filter~~
~~M1 First multiplier~~
~~M2 Second multiplier~~
~~AM Amplifier~~
~~NL1 First nonlinear circuit~~
~~NL2 Second nonlinear circuit~~
~~F1 First function generator~~
~~F2 Second function generator~~
~~G Amplification factor~~
~~GC Correction factor~~
~~NG Factor~~
~~V1, V2 Control variable~~
~~ABS Absolute value forming circuit~~
~~PK Peak value detector~~
~~TP1 Low-pass filter~~
~~TP2 Low-pass filter~~
~~TP3 Low-pass filter~~
~~S Subtractor~~
~~LIM1 Limiter~~
~~LIM2 Limiter~~
~~Xin Audio signal~~
~~Xout Output signal~~

~~Abstract of the Disclosure~~

ABSTRACT OF THE DISCLOSURE

~~So as t~~To compensate the frequency response of loudspeakers and ~~so as to give~~ provide a listener with the illusion of sonorous bass tones, a circuit is disclosed that divides ~~, it is known that~~ an audio signal ~~can be divided~~ into a first and second path, such that harmonics of the signal components with a low-frequency are generated in the second path and are admixed to the signal in the first path. To improve reproduction, especially in the bass range of weakly designed loudspeakers, the audio signal is bandpass filtered ~~(BP1)~~, weighted with a correction factor ~~GC (M1)~~, amplified with an amplification factor ~~G (AM)~~, then limited to a maximum value ~~(NL1)~~, and finally again bandpass filtered ~~(BP2)~~, in the second path ~~(P2)~~, before it is added to the original audio signal ~~(Xin)~~ in the first path ~~(P1)~~. The correction factor ~~(GC)~~ is reduced when the maximum value is exceeded, while otherwise it remains constant or is increased. Through this measure, harmonics are generated at the onset of a low-frequency audio signal, and are admixed to the original audio signal.

~~Figure 1~~

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APPARATUS AND METHOD FOR PROCESSING AN AUDIO SIGNAL TO COMPENSATE FOR THE FREQUENCY RESPONSE OF LOUDSPEAKERS

BACKGROUND OF THE INVENTION

The present invention relates generally to processing audio signals and, more particularly, to a method and apparatus for processing an audio signal to compensate for the frequency response of loudspeakers.

Devices that reproduce acoustic signals such as, for example, television sets, radio receivers, and stereo systems, include a circuit for processing the audio signals. Typically, such circuits are designed to compensate for the frequency response of the loudspeakers to improve acoustic reproduction, and to prevent overdriving the device or the system.

The most critical element in a unit for acoustic reproduction is the loudspeaker. The acoustic pressure of a loudspeaker drops about 40 db per decade below a structurally determined limit frequency. This corresponds to the transfer function of a second-order filter. On the other hand, bass reflex and transmission line loudspeakers have transfer functions which correspond to a higher order filter. Their lower limit frequency typically lies between about 50 Hz and 200 Hz. The lower the limit frequency of a loudspeaker, the more expensive it is to produce. Economical units such as, for example, television or portable radio receivers consequently are equipped with simple loudspeakers that have a relatively high lower limit frequency. To improve acoustic reproduction in the lower frequency range, the limit frequency of such units is shifted downward by preamplifying the low frequencies. However, this can cause the final amplifier and the loudspeakers to be overdriven. To prevent overdriving and possibly even destruction of the final amplifier or the loudspeaker, the output signal of the bass amplifier is fed back in such a way that

the amplification of the lower frequencies is reduced if the output signal is strong. One example of such an approach is disclosed in U.S. Patent 5,305,388.

Another conventional approach is disclosed in U.S. Patent 5,359,665. This patent describes a circuit in which the audio signal is conducted directly, via a first path, to the first input of an adder. The audio signal is simultaneously conducted, via a second path to the second input of the adder. This second path includes a low-pass filter and an amplifier with variable amplification. The output of the amplifier is fed back, through a signal level detector, to its control input. This circuit arrangement is described in the '665 patent to have the benefit of reducing overdrive of the final amplifier.

From research results of psychoacoustics, it is known that a person can still unambiguously determine the fundamental level of a tone even when only harmonics of the fundamental frequency and not the fundamental frequency itself is present in the spectrum. This psychoacoustic effect is utilized in that the harmonic of the fundamental frequency is generated and is conducted to a loudspeaker whose limit frequency lies above this fundamental frequency. A listener consequently believes that s/he hears this low fundamental frequency even though the loudspeaker does not radiate it at all.

U.S. Patent 5,668,885 describes a circuit which thereby "entices" from a loudspeaker having a relatively high lower limit frequency frequencies which are lower than its limit frequency. This is done by generating harmonics of the lower frequencies. The audio signal is conducted, via a first path, to the first input of an adder. In a second path, the audio signal passes through a low-pass filter, is rectified, passes through a second low-pass filter, is amplified, and finally is conducted to the second input of the adder.

U.S. Patent 4,150,253 likewise describes a method and a circuit, in which an audio signal

is divided into two signal paths. In the first path, the audio signal traverses a high-pass filter, so as to shift the phase based on the frequency. Those signals at the output of the high-pass filter which are at levels that exceed a given value are conducted to the input of a generator which generates the harmonics of the fundamental frequency. The level of the signals at the output of the generator is attenuated to a value below the level of the original audio signal. This attenuated signal and the original audio signal are then added together.

U.S. Patent 4,700,390 describes a so-called synthesizer, in which harmonics are generated both for the lower and higher frequencies, and are added to the original audio signal. The alleged benefit set out in the '390 patent is that this achieves better reproduction both in the low and in the high frequency range.

U.S. Patent 5,771,296 likewise describes a circuit in which the audio signal is conducted, via a first path, directly into an adder, while, via a second path, the harmonics of the lower frequencies are generated and are added in the adder to the original signal. This provides the perception to the listener that the loudspeaker radiates lower frequencies than it actually does. Finally, U.S. Patent 4,739,514 describes another circuit to improve the acoustic reproduction of low frequencies. With this circuit, too, the audio signal is conducted, via a first path, to the first input of an adder, while, via a second path consisting of an amplifier with variable amplification in series with a bandpass filter, it is conducted to the second input of the adder. A signal level detector, whose input receives the audio signal, controls the amplification of the amplifier. The above and other conventional methods and circuits have the disadvantage that, due to feedback technique employed, they react relatively slowly to rising amplitudes and, despite the feedback they tend to overdrive the device or the system. What is needed, therefore, is an audio processing technique that compensates for the frequency response of a loudspeaker without overdriving the

circuit, its components or entire reproduction system, particularly in the range of low frequencies.

SUMMARY OF THE INVENTION

Briefly, according to an aspect of the invention, a method for processing a received audio signal is disclosed. The method includes band-limiting the received audio signal to generate a first intermediate signal; multiplying the first intermediate signal by a correction factor to generate a second intermediate signal; amplifying the second intermediate signal by an amplification factor to generate a third intermediate signal; limiting the amplitude of the third intermediate signal to a specified maximum value to generate a fourth intermediate signal; band-limiting the fourth intermediate signal to generate a fifth intermediate signal; and adding the fifth intermediate signal to the received audio signal.

In another aspect of the invention, a circuit for processing an input audio signal received at an input of the circuit to provide at an output of the circuit a processed audio signal is disclosed. The circuit includes a first adder having first and second inputs and an output at which the processed audio signal is provided; a first conductive path connecting the circuit input to the first input of the first adder, the first conductive path constructed and arranged to deliver the received audio signal unaltered to the first adder; and a second conductive path connecting the circuit input to the second input of the first adder. The second conductive path includes a first bandpass filter having an output and an input connected to the circuit input; a multiplier having a first input connected to the first bandpass filter output, and a second input, and an output; a variable amplifier, having an output and an input connected to the multiplier output, for amplifying a signal received at the amplifier input in accordance with an amplification factor presented at a control

input of the amplifier; a first nonlinear circuit having an output and an input connected to the amplifier output, the nonlinear circuit limiting to a specified maximum the amplitude of a signal presented at the first nonlinear circuit input; a second bandpass filter having an input connected to the nonlinear circuit output and an output defining the circuit network output; and a first function generator having an input connected to a control output of the first nonlinear circuit, and an output connected to the multiplier second input.

In a further aspect of the invention, a circuit for processing an input audio signal received at an input of the circuit to provide at an output of the circuit a processed audio signal. The circuit band-limits the received audio signal to generate a first intermediate signal; multiplies the first intermediate signal by a correction factor to generate a second intermediate signal; amplifies the second intermediate signal by an amplification factor to generate a third intermediate signal; limits the amplitude of the third intermediate signal to a specified maximum value to generate a fourth intermediate signal; band-limits the fourth intermediate signal to generate a fifth intermediate signal; and adds the fifth intermediate signal to the received audio signal.

In a still further aspect of the invention, a circuit for processing an input audio signal received at an input of the circuit to provide at an output of the circuit a processed audio signal, is disclosed. The circuit includes a first conductive path through which the received audio signal travels; and a second conductive path through which the received audio signal travels, wherein the audio signal is processed such that harmonics of the signal components with a low-frequency are generated in the second path and are admixed to the signal in the first path, wherein in the second path the audio signal is sequentially bandpass filtered, weighted with a correction factor, amplified, limited to a maximum value, and bandpass filtered, wherein the correction factor is reduced when the maximum value is exceeded.

These and other objects, features and advantages of the present invention will become more apparent in light of the following detailed description of preferred embodiments thereof, as illustrated in the accompanying drawings.

BRIEF DESCRIPTION OF THE DRAWING

FIG. 1 is a schematic block diagram illustration of one embodiment of an audio processing circuit of the present invention;

FIG. 2 is a schematic block diagram illustration of another embodiment of an audio processing circuit of the present invention;

FIG. 3 is a schematic block diagram illustration of a further embodiment of an audio processing circuit of the present invention; and

FIG. 4 is a schematic block diagram illustration of a still further a fourth embodiment of an audio processing circuit of the present invention.

DETAILED DESCRIPTION OF THE INVENTION

The present invention is directed to an improved method and circuit for processing audio signals in a manner that compensates for the frequency response of the loudspeaker without overdriving the circuit components. Aspects of the present invention are described in detail below with reference to different embodiments of an audio processing circuit.

FIG. 1 is a schematic block diagram of one embodiment of au audio processing circuit of the present invention. In FIG. 1, an audio signal 101 is received at circuit input Xin 109. Received audio signal 101 is conducted, via a first path 105, to a first input of an adder 114 and, via a second path 107, to a second input of adder 114. Second path 107 includes a first bandpass

filter 102, a multiplier 104, an amplifier 106 with variable amplification, a nonlinear circuit 108, and a second bandpass filter 112, all connected in series. The processed audio signal 103 is available at circuit output node Xout 199. The circuit output 199 is directly connected to the output of adder 114. A control output of nonlinear circuit 108 is connected to the input of a function generator 110. Nonlinear circuit 108 generates at its control output a control variable ("V") 118 which generates a correction factor ("GC") 116 based on the value of control variable 118. The output of function generator 110 is connected to multiplier, 104 providing correction factor ("GC") 116 to multiplier 104. As shown in Figure 1, a control variable ("G") 120 is applied to a control input of amplifier 106.

During operation, the received audio signal 101 is band-limited by bandpass filter 102. The audio signal is then multiplied by variable correction factor GC 116 in multiplier 104. The resulting product provided at the output of multiplier 104 is amplified in amplifier 106 by amplification factor G 120. Nonlinear circuit 108 limits the amplitude of the audio signal delivered by amplifier 106 to a specified value. Embodiments of nonlinear circuit 108 are described in detail below. The output signal of nonlinear circuit 108 is band-limited by bandpass filter 112. As noted, nonlinear circuit 108 generates control variable V 118 the value of which is used by function generator 110 to generate correction factor GC 116. Correction factor GC 116 is varied by function generator 110 based on the value of control variable 118. The value of correction factor GC 116 is reduced by function generator 110 when control variable 118 is greater than a predetermined maximum value, a condition that can result in an overdrive condition. On the other hand, if the level of the audio signal lies within allowed limits, correction factor GC 116 is increased by function generator 110. Preferably, the value of correction factor GC 116 is limited to a maximum value of one (1). This is described in greater detail below.

A schematic block diagram of another embodiment of the audio processing circuit of the present invention, referred to as audio processing circuit 200, is depicted in FIG. 2. Audio processing circuit 200 will now be described in conjunction with the embodiment of audio processing circuit 100 described above and illustrated in FIG. 1.

As shown in FIG. 2, the arrangement of the components of audio processing circuit 200 is similar to that of audio processing circuit 100. However, in the exemplary embodiment illustrated in Figure 2, function generator 110 is implemented as a low-pass filter. This function generator is referred to in FIG. 2 as low-pass filter 210. Also, in this embodiment, nonlinear circuit 108 is implemented as a limiter circuit which cuts off the signal amplitude above a specified threshold. This embodiment of nonlinear circuit 108 is referred to in FIG. 2 as limiter circuit 204. It should be understood to those of ordinary skill in the art that the above-noted components of audio processing circuit 100 and audio processing circuit 200 can be implemented with any well-known circuit components now or later developed.

If the amplitude of the audio signal amplified by amplifier 106 exceeds the specified threshold, nonlinear circuit 108, 204 sets the value of control variable V 118 to a negative value. On the other hand, when the amplitude of the signal lies below the specified threshold, nonlinear circuit 108, 204 generates a control variable V 118 having a positive value. As noted, control variable V 118 is received by low-pass filter 210 which generates correction factor 116 based on the value of control variable 118. In accordance with the embodiment illustrated in FIG. 2, correction factor GC 116 for multiplier 104 is created by filtering control variable V 118 by low-pass filter 210.

The nonlinear operation in nonlinear circuit 108, which limits the amplitudes of the audio signal to a specified threshold, generates audio signals with lower frequencies, which are also

called harmonics of the bass signal. The shape of these harmonics is determined by the selection of the nonlinear operation implemented in nonlinear circuit 108 and by the dimensioning of bandpass filter 112. A useful form of these harmonics can be determined, for example, by calculation or by experiment, so as to make the beginning of an audio signal with low frequencies, such as the striking of a drum, appear clearer and brighter to a listener. The choice of function implemented in function generator 110, 210 determines the time which passes between the beginning of a strong, low-frequency tone and the reduction of correction factor GC 116 to such an extent that nonlinear circuit 108 no longer generates harmonics. The length of this time interval, which is regarded as a time constant, is determined by the dimensioning of low-pass filter 210 and the choice of the positive and negative values of control variable V 118.

One advantage of audio processing circuits 100, 200 is that, with a small signal amplitude, amplifier 106 operates at full amplification and thus partially compensates the frequency characteristic of a loudspeaker. On the other hand, if the signal amplitude is sufficiently large, the frequency characteristic of the loudspeaker can be slightly compensated to avoid the loudspeaker from being overdriven. Thus, upon the onset of a bass signal, the bass signal is enriched with harmonics so that a listener, despite the lack of bass volume from the loudspeaker, has the sensation of clearly and loudly hearing the bass frequencies.

A further embodiment of the audio processing circuit of the present invention, referred to as audio processing circuit 300, is shown in FIG. 3. Audio processing circuit 300 will now be described in conjunction with the embodiments of audio processing circuits 100 and 200 described above and illustrated in FIGs. 1 and 2, respectively.

In the exemplary embodiment illustrated in FIG. 3, a more detailed illustration of one embodiment of nonlinear circuit 108 is shown. This embodiment of nonlinear circuit 108 is

composed of a nonlinear circuit 302 and a function generator 304. The audio signal provided to the input of nonlinear circuit 108 is, as noted, the output signal of amplifier 106. This signal is directly conducted to the input of nonlinear circuit 302 and to the input of function generator 304. The output of function generator 304 is connected to a control input of nonlinear circuit 302. The signal output of nonlinear circuit 302 is connected to the input of bandpass filter 112, while a control output of nonlinear circuit 302 is connected to a function generator 310. Function generator 310 can be implemented as function generator 110 described above with reference to FIG. 1, or as low-pass filter 210 described above with reference to FIG. 2.

Nonlinear circuit 302 continuously generates harmonics of the low-frequency components of the audio signal, which are weighted with variable factor ("NG") 306 by function generator 304. The factor NG 306 is a function of the signal input to function generator 304. Depending on the choice of function for function generator 304, manifold acoustic effects can be created. For example, function generator 304 can be designed so that nonlinear circuit 302 more strongly generates harmonics as soon as the signal amplitude is to be limited, so as to prevent overdrive. In this way, signal energy is distributed among the higher frequency harmonics, which a loudspeaker or a loudspeaker system can more accurately reproduce. Although the energy of the lower frequency signal components is reduced in this approach, the listener nevertheless has the impression of a full bass sound due to the above-mentioned psychoacoustic effects.

A still further embodiment of the audio processing circuit of the present invention, referred to as audio processing circuit 400, is shown in FIG. 4. Audio processing circuit 400 will now be described in conjunction with the embodiments of audio processing circuit 100, 200 and 300 described above and illustrated in FIGs. 1, 2 and 3, respectively. In this exemplary illustration of audio processing circuit 400, one embodiment of a detailed implementation of nonlinear circuit

308 is illustrated. This illustrative embodiment is referred to in FIG. 4 as nonlinear circuit 408. Nonlinear circuit 408 includes embodiments of nonlinear circuit 302, referred to as nonlinear circuit 402 herein, and function generator 304, referred to herein as function generator 404.

In this embodiment, the signal input to nonlinear circuit 408 which, as noted, is the output signal from amplifier 106, is conducted to a first input of an adder 402, to the input of an absolute value forming circuit 404, and to the input of a peak value detector 406. The output of peak value detector 406 is connected to the input of a low-pass filter 410 and a low-pass filter 412. The output of low-pass filter 410 is connected to a first input of a subtractor 414, and the output of low-pass filter 412 is connected to a second input of subtractor 414. The output of subtractor 414 is connected, via a limiter 416, to the first input of a multiplier 418.

The output of absolute value forming circuit 404 is connected to the second input of multiplier 418, the output of which is connected to a second input of adder 402. The output of adder 402 is connected to the input of a limiter 420. The control output of limiter 420 is provided to function generator 410. As with function generator 310, function generator 410 can be implemented as function generator 110 or as low-pass filter 210. The output of limiter 420 is connected to the input of bandpass filter 112. The processed audio signal 103 for a loudspeaker or a loudspeaker system is available at the output of bandpass filter 112.

Peak value detector 406 determines the level of the maximum amplitude occurring during a specified time interval T. The output signal of peak value detector 406 is time-averaged by the two low-pass filters 410 and 412. In one embodiment, the time constant of low-pass filter 412 is smaller than that of low-pass filter 410. That is, the cut-off frequency of low-pass filter 412 with the smaller time constant is higher than that of low-pass filter 410 with the larger time constant. Because of the smaller time constant, the output signal of low-pass filter 412 follows a change of

the input signal faster than does the output signal of low-pass filter 412. The absolute value forming circuit 404 forms the absolute value of the input signal, which is weighted in multiplier 418 by factor NG 306 generated by subtractor 416. Limiter 416 limits factor NG 306 to a range between 0 and 1. The weighted absolute value of the input signal is added in adder 402 to the input signal, and the resulting sum is limited to a specified amplitude by limiter 420 so as to prevent overdrive.

For example, if the amplitude of the input signal rises discontinuously, the level at the output of low-pass filter 412 will rise faster, due to its smaller time constant, than at the output of low-pass filter 410. As a result, factor NG 306, which is to be regarded as a control variable, assumes a positive value for rising amplitudes in the input signal. As the rate at which the amplitude of the input signal rises increases, more harmonics will be generated and will be admixed to the input signal. On the other hand, if the amplitude falls, factor NG 306 becomes negative, because now the level at the output of low-pass filter 412, due to its smaller time constant, becomes smaller than the level at the output of low-pass filter 410. Because the factor NG 306 has a lower limit of zero, no harmonics are admixed to the audio signal when the amplitudes are falling.

A significant advantage of the invention is that the nonlinear operation of nonlinear circuit 408, and the function of function generator 410, determine the form of the harmonics as well as the time of their generation. It should be understood by those of ordinary skill in the art that adjustments in the nonlinear operation of the nonlinear circuit 108, 208, 308, 408 and of the function of function generator110, 210, 310, 410, the invention can easily be adapted to loudspeakers with different characteristics, so that optimum compensation of the frequency response of a loudspeaker is always achieved. Because the amplitude of the audio signal is limited

to a specified value by nonlinear circuit, the inventive circuit reacts much faster than the prior art to rising amplitudes of the audio signal.

The invention is especially suited for acoustic reproduction units, e.g. television units, portable radios, which are equipped with loudspeakers with a weak bass range, because the invention prevents overdriving the entire reproduction system and at the same time offers the listener the illusion of sonorous basses, even though the loudspeakers really do not radiate these low bass frequencies.

Although the present invention has been shown and described with respect to several preferred embodiments thereof, various changes, omissions and additions to the form and detail thereof, may be made therein, without departing from the spirit and scope of the invention.

What is claimed is:

Clean Copy of the Claims

Following Entry of This Amendment

--13. (New) A method for processing a received audio signal comprising:

band-limiting the received audio signal to generate a first intermediate signal;

multiplying said first intermediate signal by a correction factor to generate a second intermediate signal;

amplifying said second intermediate signal by an amplification factor to generate a third intermediate signal;

limiting the amplitude of said third intermediate signal to a specified maximum value to generate a fourth intermediate signal;

band-limiting said fourth intermediate signal to generate a fifth intermediate signal; and

adding said fifth intermediate signal to said received audio signal.--

--14. (New) The method of claim 13, further comprising:

adjusting said correction factor based on whether said third intermediate signal exceeds a predetermined threshold value.--

--15. (New) The method of claim 14, wherein said adjusting said correction factor comprises:

reducing said correction factor when said third intermediate signal exceeds said predetermined threshold value.--

--16. (New) The method of claim 15, wherein said adjusting said correction factor comprises:

increasing said correction factor when said third intermediate signal does not exceed said predetermined threshold value.--

--17. (New) The method of claim 14, wherein said adjusting said correction factor comprises:

generating a control variable based on said amplitude of said third intermediate signal; and

generating said correction factor as a function of said control variable.--

--18. (New) The method of claim 17, wherein said generating a correction factor as a function of said control variable is performed by a low-pass filter.--

--19. (New) The method of claim 13, wherein said limiting the amplitude of said third intermediate signal to a specified maximum value comprises:

generating harmonics of low-frequency signal components of said received audio signal; and

weighting said harmonics with a variable factor.--

--20. (New) The method of claim 19, wherein said weighting said harmonics with a variable factor comprises:

generating said variable factor as a function of said third intermediate signal.--

--21. (New) The method of claim 20, wherein said step of generating harmonics comprises:

increasingly generating harmonics at the beginning of a low-frequency signal.--

--22. (New) The method of claim 20, wherein said generating said variable factor as a function of said third intermediate signal comprises:

detecting a peak value of said third intermediate signal in accordance with a predetermined

function of said third intermediate signal to generate a sixth intermediate signal;

low-pass filtering said sixth intermediate signal separately with first and second time constants; and

generating a difference signal of the two low-pass filtered signals, wherein said difference signal is generated as said variable factor.--

--23. (New) The method of claim 20, wherein said weighting further comprises the steps of:

determining an absolute value of said third intermediate signal;

multiplying said absolute value of said third intermediate signal with said variable factor to generate a seventh intermediate signal;

adding to said third intermediate signal to said seventh intermediate signal to form an eight intermediate value; and

limiting amplitudes of said eight intermediate value to a specified value.--

--24. (New) A circuit for processing an input audio signal received at an input of the circuit to provide at an output of the circuit a processed audio signal, the circuit comprising:

a first adder having first and second inputs and an output at which the processed audio signal is provided;

a first conductive path connecting the circuit input to said first input of said first adder, said first conductive path constructed and arranged to deliver said received audio signal unaltered to said first adder;

a second conductive path connecting said circuit input to said second input of said first adder, said second conductive path comprising,

a first bandpass filter having an output and an input connected to said circuit input;

a multiplier having a first input connected to said first bandpass filter output, and a second input, and an output;

a variable amplifier, having an output and an input connected to said multiplier output, for amplifying a signal received at said amplifier input in accordance with an amplification factor presented at a control input of said amplifier;

a first nonlinear circuit having an output and an input connected to said amplifier output, said nonlinear circuit limiting to a specified maximum the amplitude of a signal presented at said first nonlinear circuit input; and

a second bandpass filter having an input connected to said nonlinear circuit output and an output defining said circuit network output; and

a first function generator having an input connected to a control output of said first nonlinear circuit, and an output connected to said multiplier second input.--

--25. (New) The circuit arrangement of claim 24, wherein said first function generator comprises a first low-pass filter.--

--26. (New) The circuit arrangement of claim 24, wherein said first nonlinear circuit comprises:

a second nonlinear circuit having an input and output connected to said input and output, respectively, of said first nonlinear circuit, a control output defining said control output of said first nonlinear circuit, and a control input to which said second nonlinear circuit is responsive; and

a second function generator having an input connected to said input of said first nonlinear circuit and an output connected to said control input of said second nonlinear circuit.--

--27. (New) The circuit arrangement of claim 26, wherein said second function generator comprises:

a peak value detector circuit having an output and an input connected to said second function generator input,

a second low-pass filter having an output and an input connected to said peak value detector output;

a third low-pass filter having an output and an input connected to said peak value detector output;

a subtractor having first and second inputs connected to said outputs of said second and third low-pass filters, respectively, and an output; and

a first limiter circuit having an input connected to said subtractor output, and an output connected to said second function generator control input.--

--28. (New) The circuit arrangement of claim 27, wherein said second nonlinear circuit comprises:

an absolute value forming circuit having an output and an input connected to said first nonlinear circuit input;

a second multiplier having a first input connected to said first limiter circuit output and a second input connected to said absolute value forming circuit output;

a second adder having an output, a first input connected to said first nonlinear circuit input, and a second input connected to said second multiplier output; and

a second limiter circuit having an input connected to said second adder output, a control

output connected to said first function generator, and an output connected to said second bandpass filter input.--

--29. (New) A circuit for processing an input audio signal received at an input of the circuit to provide at an output of the circuit a processed audio signal, the circuit comprising:

means for band-limiting the received audio signal to generate a first intermediate signal;

means for multiplying the first intermediate signal by a correction factor to generate a second intermediate signal;

means for amplifying the second intermediate signal by an amplification factor to generate a third intermediate signal;

means for limiting the amplitude of said third intermediate signal to a specified maximum value to generate a fourth intermediate signal;

means for band-limiting said fourth intermediate signal to generate a fifth intermediate signal; and

means for adding said fifth intermediate signal to said received audio signal.--

--30. (New) The circuit of claim 29, further comprising:

means for adjusting said correction factor based on whether said third intermediate signal exceeds a predetermined threshold value.--

--31. (New) The circuit of claim 30, wherein said adjusting means comprises:

means for reducing said correction factor when said third intermediate signal exceeds said predetermined threshold value, and for increasing said correction factor when said third

intermediate signal does not exceed said predetermined threshold value.--

--32. (New) A circuit for processing an input audio signal received at an input of the circuit to provide at an output of the circuit a processed audio signal, the circuit comprising:

a first conductive path through which the received audio signal travels;

a second conductive path through which the received audio signal travels, wherein the audio signal is processed such that harmonics of the signal components with a low-frequency are generated in the second path and are admixed to the signal in the first path, wherein in the second path the audio signal is sequentially bandpass filtered, weighted with a correction factor, amplified, limited to a maximum value, and bandpass filtered, wherein the correction factor is reduced when the maximum value is exceeded.--

REMARKS

Claims 1-12 have been canceled. Claims 13-32 have been added. Claims 13-32 remain.

The specification has been amended following the translation of the application to English.

Examination on the merits is respectfully requested.

If a telephone interview could assist in the prosecution of this application, please call the undersigned attorney.

Respectfully submitted,

Patrick O'Shea

Patrick J. O'Shea
Registration No. 35,305
Samuels, Gauthier & Stevens LLP
225 Franklin Street, Suite 3300
Boston, Massachusetts 02110
Telephone: (617) 426-9180 x121

VERSION WITH MARKINGS TO SHOW CHANGES MADE TO CLAIMS

Please cancel claims 1-12, and add claims 13-32 as follows:

1. (Canceled) A method for processing an audio signal (Xin), which, via a first path (P1), is conducted to the first input of a first adder (A1), characterized in that the audio signal (Xin) is band-limited in a second path (P2) by means of a first bandpass filter (BP1) , and that, at the output of the first bandpass filter (BP1), the band-limited audio signal is multiplied by a correction factor (GC), is amplified by an amplification factor (G), and then its amplitude is limited to a specified maximum by means of a nonlinear circuit (NL1), and that the correction factor (GC) is reduced when the prescribed maximum is exceeded, but otherwise remains constant or is increased, and that the audio signal at the output of the first nonlinear circuit (NL1) is band-limited by means of a second bandpass filter (BP2), and that the band-limited audio signal at the output of the second bandpass filter (BP2) is added, in the adder (AD), to the audio signal of the first path (P1).

2. (Canceled)The method of Claim 1, characterized in that the correction factor (GC) is a function (F1) of a control variable (V), which is generated by the first nonlinear circuit (NL1).

3. (Canceled)The method of Claim 2, characterized in that the correction factor (GC) assumes a first value (V1) if the amplitude is less than the prescribed threshold, and assumes a second value (V2) if the amplitude is greater than the specified threshold.

4. (Canceled)The method of Claim 2 or 3, characterized in that the function (F1) is implemented by means of a first low-pass filter (TP1).

5. (Canceled) The method of one of the Claims 1 to 4, characterized in that the nonlinear circuit (NL1) generates harmonics of the low-frequency signal components and weights them with a variable factor (NG).

6. (Canceled) The method of Claim 5, characterized in that the variable factor (NG) is a function (F2) of the input signal of the first nonlinear circuit (NL1).

7. (Canceled) The method of Claim 6, characterized in that the first nonlinear circuit (NL1) increasingly generates harmonics at the beginning of a low-frequency signal.

8. (Canceled) The method of Claim 7, characterized in that the input signal of the first nonlinear circuit (NL1) is detected by a peak value detector (PK) in the first nonlinear circuit (NL1), in accordance with the function (F2) of the input signal of the first nonlinear circuit (NL1), and that the output signal of said peak value detector is low-pass filtered in a second and third low-pass filter (TP2, TP3) with different time constants, and that the difference signal of the two low-pass filtered signals represents the variable factor (NG), and that the absolute value of the input signal of the first nonlinear circuit (NL1) is weighted with this factor (NG) and is added to the input signal of the first nonlinear circuit (NL1) to form a summation value, whose amplitude is limited to a specified value.

9. (Canceled) A circuit arrangement to process an audio signal (Xin) which, via a first path (P1), is conducted to the first input of a first adder (A1), whose output delivers the output signal (Xout), characterized in that the audio signal (Xin) is conducted to the second input of the first adder (A1) via a second path (P2), consisting of a first bandpass filter (BP1), a first multiplier (M1), an amplifier (AM) with variable amplification, a first nonlinear circuit (NL1), and a second bandpass filter (BP2), all connected in series, and that a control output of the first nonlinear circuit (NL1) is connected to the input of a first function generator (F1), whose output is connected to the first multiplier (M1), and that an amplification (G) is applied to the control input of the amplifier (AM).

10. (Canceled) The circuit arrangement of Claim 9, characterized in that the first function generator (F1) is designed as a first low-pass filter (TP1).

11. (Canceled) The circuit arrangement of Claim 9 or 10, characterized in that the first nonlinear circuit (NL1) consists of a second nonlinear circuit (NL2) and the second function generator (F2), and that the input signal of the first nonlinear circuit (NL1) is applied to the input of the second nonlinear circuit (NL2) and to the input of the second function generator (F2), whose output is connected to the control input of the second nonlinear circuit (NL2), and that the control output of the second nonlinear circuit (NL2) is connected to the first function generator (F1) or to the first low-pass filter (TP1), and that the signal output of the second nonlinear circuit (NL2) is connected to the input of the second bandpass filter (BP2).

12. (Canceled) The circuit arrangement of Claim 11, characterized in that the second function generator (F2) consists of a peak value detector (PK), a second and a third low-pass filter (TP2, TP3), a first limiter (LIM1), and a subtractor (S), and that the second nonlinear circuit (NL2) consists of an absolute value forming circuit (ABS), a second multiplier (M2), a second adder (A2), and a second limiter (LIM2), and that the input signal of the first nonlinear circuit (NL1) is conducted to the first input of the second adder (A2), to the input of the absolute value forming circuit (ABS), and to the input of the peak value detector (PK), whose output is connected to the input of the second low-pass filter (TP2) and to the input of the third low-pass filter (TP3), and that the output of the second low-pass filter (TP2) is connected to the first input of a subtractor (S), and the output of the third low-pass filter (TP3) is connected to the second input of a subtractors (S), whose output is connected, via the first limiter (LIM1), to the first input of the second multiplier (M2), and that the output of the absolute value forming circuit (ABS) is connected to the second input of the second multiplier (M2), whose output is connected to the second input of the second adder (A2), and that the output of the second adder (A2) is connected to the input of the second limiter (LIM2), whose control output is connected to the first function generator (F1) or to the first low-pass filter (TP1), and whose output is connected to the input of the second bandpass filter (BP2).

--13. (New) A method for processing a received audio signal comprising:

band-limiting the received audio signal to generate a first intermediate signal;

multiplying said first intermediate signal by a correction factor to generate a second intermediate signal;

amplifying said second intermediate signal by an amplification factor to generate a third

intermediate signal;

limiting the amplitude of said third intermediate signal to a specified maximum value to generate a fourth intermediate signal;

band-limiting said fourth intermediate signal to generate a fifth intermediate signal; and

adding said fifth intermediate signal to said received audio signal.--

--14. (New) The method of claim 13, further comprising:

adjusting said correction factor based on whether said third intermediate signal exceeds a predetermined threshold value.--

--15. (New) The method of claim 14, wherein said adjusting said correction factor comprises:

reducing said correction factor when said third intermediate signal exceeds said predetermined threshold value.--

--16. (New) The method of claim 15, wherein said adjusting said correction factor comprises:

increasing said correction factor when said third intermediate signal does not exceed said predetermined threshold value.--

--17. (New) The method of claim 14, wherein said adjusting said correction factor comprises:

generating a control variable based on said amplitude of said third intermediate signal; and

generating said correction factor as a function of said control variable.--

--18. (New) The method of claim 17, wherein said generating a correction factor as a function of

said control variable is performed by a low-pass filter.--

--19. (New) The method of claim 13, wherein said limiting the amplitude of said third intermediate signal to a specified maximum value comprises:

generating harmonics of low-frequency signal components of said received audio signal; and

weighting said harmonics with a variable factor.--

--20. (New) The method of claim 19, wherein said weighting said harmonics with a variable factor comprises:

generating said variable factor as a function of said third intermediate signal.--

--21. (New) The method of claim 20, wherein said step of generating harmonics comprises:

increasingly generating harmonics at the beginning of a low-frequency signal.--

--22. (New) The method of claim 20, wherein said generating said variable factor as a function of said third intermediate signal comprises:

detecting a peak value of said third intermediate signal in accordance with a predetermined function of said third intermediate signal to generate a sixth intermediate signal;

low-pass filtering said sixth intermediate signal separately with first and second time constants; and

generating a difference signal of the two low-pass filtered signals, wherein said difference signal is generated as said variable factor.--

--23. (New) The method of claim 20, wherein said weighting further comprises the steps of:

determining an absolute value of said third intermediate signal;

multiplying said absolute value of said third intermediate signal with said variable factor to generate a seventh intermediate signal;

adding to said third intermediate signal to said seventh intermediate signal to form an eight intermediate value; and

limiting amplitudes of said eight intermediate value to a specified value.--

--24. (New) A circuit for processing an input audio signal received at an input of the circuit to provide at an output of the circuit a processed audio signal, the circuit comprising:

a first adder having first and second inputs and an output at which the processed audio signal is provided;

a first conductive path connecting the circuit input to said first input of said first adder, said first conductive path constructed and arranged to deliver said received audio signal unaltered to said first adder;

a second conductive path connecting said circuit input to said second input of said first adder, said second conductive path comprising,

a first bandpass filter having an output and an input connected to said circuit input;

a multiplier having a first input connected to said first bandpass filter output, and a second input, and an output;

a variable amplifier, having an output and an input connected to said multiplier output, for amplifying a signal received at said amplifier input in accordance with an

amplification factor presented at a control input of said amplifier;

a first nonlinear circuit having an output and an input connected to said amplifier output, said nonlinear circuit limiting to a specified maximum the amplitude of a signal presented at said first nonlinear circuit input; and

a second bandpass filter having an input connected to said nonlinear circuit output and an output defining said circuit network output; and

a first function generator having an input connected to a control output of said first nonlinear circuit, and an output connected to said multiplier second input.--

--25. (New) The circuit arrangement of claim 24, wherein said first function generator comprises a first low-pass filter.--

--26. (New) The circuit arrangement of claim 24, wherein said first nonlinear circuit comprises:

a second nonlinear circuit having an input and output connected to said input and output, respectively, of said first nonlinear circuit, a control output defining said control output of said first nonlinear circuit, and a control input to which said second nonlinear circuit is responsive; and

a second function generator having an input connected to said input of said first nonlinear circuit and an output connected to said control input of said second nonlinear circuit.--

--27. (New) The circuit arrangement of claim 26, wherein said second function generator comprises:

a peak value detector circuit having an output and an input connected to said second function generator input,

a second low-pass filter having an output and an input connected to said peak value detector output;

a third low-pass filter having an output and an input connected to said peak value detector output;

a subtractor having first and second inputs connected to said outputs of said second and third low-pass filters, respectively, and an output; and

a first limiter circuit having an input connected to said subtractor output, and an output connected to said second function generator control input.--

--28. (New) The circuit arrangement of claim 27, wherein said second nonlinear circuit comprises:

an absolute value forming circuit having an output and an input connected to said first nonlinear circuit input;

a second multiplier having a first input connected to said first limiter circuit output and a second input connected to said absolute value forming circuit output;

a second adder having an output, a first input connected to said first nonlinear circuit input, and a second input connected to said second multiplier output; and

a second limiter circuit having an input connected to said second adder output, a control output connected to said first function generator, and an output connected to said second bandpass filter input.--

--29. (New) A circuit for processing an input audio signal received at an input of the circuit to provide at an output of the circuit a processed audio signal, the circuit comprising:

means for band-limiting the received audio signal to generate a first intermediate signal;

means for multiplying the first intermediate signal by a correction factor to generate a second intermediate signal;

means for amplifying the second intermediate signal by an amplification factor to generate a third intermediate signal;

means for limiting the amplitude of said third intermediate signal to a specified maximum value to generate a fourth intermediate signal;

means for band-limiting said fourth intermediate signal to generate a fifth intermediate signal; and

means for adding said fifth intermediate signal to said received audio signal.--

--30. (New) The circuit of claim 29, further comprising:

means for adjusting said correction factor based on whether said third intermediate signal exceeds a predetermined threshold value.--

--31. (New) The circuit of claim 30, wherein said adjusting means comprises:

means for reducing said correction factor when said third intermediate signal exceeds said predetermined threshold value, and for increasing said correction factor when said third intermediate signal does not exceed said predetermined threshold value.--

--32. (New) A circuit for processing an input audio signal received at an input of the circuit to provide at an output of the circuit a processed audio signal, the circuit comprising:

a first conductive path through which the received audio signal travels;

a second conductive path through which the received audio signal travels, wherein the audio signal is processed such that harmonics of the signal components with a low-frequency are generated in the second path and are admixed to the signal in the first path, wherein in the second path the audio signal is sequentially bandpass filtered, weighted with a correction factor, amplified, limited to a maximum value, and bandpass filtered, wherein the correction factor is reduced when the maximum value is exceeded.--

3/prts

Description

A Method for Processing an Audio Signal

The invention relates to a method and to a circuit for processing an audio signal which is applied, via a first path, to the first input of an adder.

Such methods and circuits are used in devices for acoustic reproduction, such as e.g. television sets, radio receivers, or stereo systems, to compensate the frequency response of the loudspeakers to improve acoustic reproduction, and to prevent overdriving the device or the system.

The most critical element in a unit for acoustic reproduction is the loudspeaker, whose acoustic pressure drops about 40 db per decade below a structurally determined limit frequency. This corresponds to the transfer function of a second-order filter. On the other hand, bass reflex and transmission line loudspeakers have transfer functions which correspond to a higher order filter. Their lower limit frequency typically lies between about 50 Hz and 200 Hz. The lower the limit frequency of a loudspeaker, the more expensive is it to produce. Economical units, such as e.g. television or portable radio receivers consequently are equipped with simpler loudspeakers, whose lower limit frequency is relatively high. To improve acoustic reproduction in the lower frequency range, the limit frequency is shifted downward by preamplifying the low frequencies. However, this can cause the final amplifier and the loudspeakers to be overdriven. To prevent overdriving and possibly even destruction of the final amplifier or the loudspeaker, the output signal of the bass amplifier is fed back in such a way that the amplification of the lower frequencies is reduced if the output signal is strong. Such a method is known from the US-PS 5,305,388.

The US-PS 5,359,665 describes a circuit in which the audio signal is conducted directly, via a first path, to the first input of an adder, while at the same time it is conducted, via a second path, and via a low-pass filter and an amplifier with variable amplification, to the second input of the adder. The output of the amplifier is fed back, through a signal level detector, to its control input. This procedure reduces overdrive of the final amplifier.

From research results of psychoacoustics, it is known that a person can still unambiguously determine the fundamental tone level of a tone even if the fundamental frequency is not even present in the spectrum, but only harmonics of the fundamental. This psychoacoustic effect is utilized in that the harmonic of the fundamental frequency is generated and is conducted to a loudspeaker whose limit frequency lies above this fundamental frequency. A listener consequently believes that he hears this low fundamental frequency even though the loudspeaker does not radiate it at all.

The US-PS 5,668,885 describes a circuit which thereby "entices" from a loudspeaker with a relatively high lower limit frequency still lower frequencies than its limit frequency. This is done by generating harmonics of the lower frequencies. The audio signal is conducted, via a first path, to the first input of an adder. In a second path, the audio signal passes through a low-pass filter, is rectified, once again passes through a low-pass filter, is amplified, and finally is conducted to the second input of the adder.

The US-PS 4,150,253 likewise describes a method and a circuit, in which an audio signal is divided into two signal paths. In the first path, the audio signal traverses a high-pass filter, so as to shift the phase in dependence on the frequency. Those signals at the output of the high-pass filter, whose levels exceed a given value, are conducted to the input of a generator to generate the harmonics of the fundamental frequency. The level of the signals at the output of the generator is attenuated to a value below the level of the original audio signal. This attenuated signal and the original audio signal are added together.

The US-PS 4,700,390 describes a so-called synthesizer, in which harmonics are generated both for the lower and higher frequencies, and are added to the original audio signal. This is supposed to achieve better reproduction both in the low and in the high frequency range.

The US-PS 5,771,296 likewise describes a circuit in which the audio signal is conducted, via a first path, directly into an adder, while, via a second path, the harmonics of the lower frequencies are generated and are added in the adder to the original signal, so as to make the listener believe that a loudspeaker radiates lower frequencies than it actually does.

Finally, the US-PS 4,739,514 describes another circuit to improve the acoustic reproduction of low frequencies. With this circuit, too, the audio signal is conducted, via a first path, to the first input of an adder, while, via a second path consisting of an amplifier with variable amplification in series with a bandpass filter, it is conducted to the second input of the adder. A signal level detector, whose input receives the audio signal, controls the amplification of the amplifier.

All the referenced, known methods and circuits have the disadvantage that, due to feedback, they react relatively slowly to rising amplitudes and, despite the feedback they tend to overdrive.
It is therefore the object of the invention to design a method in accordance with the generic part of Claim 1 and a circuit in accordance with the generic part of Claim 9, in such a fashion that the frequency response of a loudspeaker is compensated, its acoustic reproduction is improved, and overdrive of the entire reproduction system is prevented, especially in the range of low frequencies.

The invention achieves this object in terms of method in that the audio signal is band-limited in a second path by means of a first bandpass filter , and that, at the output of the first bandpass filter, the band-limited audio signal is multiplied by a correction factor, is amplified by an amplification factor, and then its amplitude is limited to a specified maximum by means of a nonlinear circuit, and that the correction factor is reduced when the prescribed maximum is exceeded, but otherwise remains constant or is increased, and that the audio signal at the output of the first nonlinear circuit is band-limited by means of a second bandpass filter, and that the band-limited audio signal at the output of the second bandpass filter is added, in the adder, to the audio signal of the first path.

The invention achieves this object in terms of apparatus in that the audio signal is conducted to the second input of the first adder via a second path, consisting of a first bandpass filter, a first multiplier, an amplifier with variable amplification, a first nonlinear circuit, and a second bandpass filter, all connected in series, and that a control output of the first nonlinear circuit is connected to the input of a first function generator, whose output is connected to the first multiplier, and that an amplification is applied to the control input of the amplifier.

The invention will now be described and explained in terms of the inventive embodiments shown in the figures.

Figure 1 shows a first embodiment of the invention.

Figure 2 shows a second embodiment of the invention.

Figure 3 shows a third embodiment of the invention.

Figure 4 shows a fourth embodiment of the invention.

The inventive method will be described and explained in terms of the first embodiment of the inventive circuit, shown in Figure 1.

In Figure 1, an audio signal Xin is conducted, via a first path P1, to the first input of a first adder A1 and, via a second path P2, to the second input of the adder A1. The path P2 consists of a first bandpass filter BP1, a first multiplier M1, an amplifier AM with variable amplification, a first nonlinear circuit NL1, and a second bandpass filter BP2 all connected in series. The output audio signal Xout can be taken from the output of the adder A1. A control output of the first nonlinear circuit NL1 is connected to the input of a function generator F1. The output of the function generator F1 is connected to the first multiplier M1. A control variable G is applied to the control input of the amplifier AM.

The audio signal Xin is band-limited by the first bandpass filter BP1. Then the audio signal is multiplied by a variable correction factor GC in the multiplier M1. The product at the output of the multiplier M1 is amplified in the amplifier AM by the amplification factor G. The nonlinear circuit NL1 limits the amplitude of the audio signal delivered by the amplifier AM to a specified value. The output signal of the nonlinear circuit NL1 is band-limited by means of the bandpass filter BP2. The nonlinear circuit NL1 creates a control variable V, from which the function generator F1 generates the correction factor GC. This correction factor GC is varied by the function generator F1 in dependence on the control variables V, in such a way that it is reduced in the case of overdrive. On the other hand, if the level of the audio signal lies within its allowed limits, the correction factor GC is increased by the function generator F1, but at most up to the value 1.

The second embodiment of the invention, shown in Figure 2, will now be described and will be explained in conjunction with the first embodiment of Figure 1.

In the second embodiment in Figure 2, the function generator F1 is implemented as a low-pass filter TP1, and the first nonlinear circuit NL1 is implemented as a limiter, which cuts off the signal amplitude above a specified threshold.

If the signal amplitude exceeds the specified threshold, the nonlinear circuit NL1 conducts the control variable V with negative value V1 to the low-pass filter TP1. If the signal amplitude lies below the specified threshold, the nonlinear circuit NL1 generates a control variable V with a positive value V2. The correction factor GC for the multiplier M1 is created by filtering the control variable V by means of the low-pass filter TP1.

The nonlinear operation in the nonlinear circuit NL1, which limits the amplitudes of the audio signal to a specified threshold, generates audio signals with the lower frequencies, which are also called harmonics of the bass signal. The shape of these harmonics is determined by the choice of the nonlinear operation in the nonlinear circuit NL1 and by the dimensioning of the bandpass filter BP2. A useful form of these harmonics can be determined e.g. by calculation or by experiment, so as to make the beginning of an audio signal with low frequencies, e.g. the striking of a drum, appear clearer and brighter to a listener. The choice of function of the function generator F1 determines the time which passes between the beginning of a strong, low-frequency tone and the reduction of the correction factor GC to such an extent that the nonlinear circuit NL1 no longer generates harmonics. The length of this time interval, which is regarded as a time constant, is determined by the dimensioning of the low-pass filter TP1 and the choice of the two control variables V1 and V2.

The inventive methods described above achieve the following effects:

With a small signal amplitude, the amplifier operates at full amplification and thus partially compensates the frequency characteristic of a loudspeaker. On the other hand, if the signal amplitude is sufficiently large, the frequency

characteristic of the loudspeaker can be compensated only slightly, because otherwise it would be overdriven. Upon the onset of a bass signal, this bass signal is enriched with harmonics, so that a listener, despite the lack of bass volume from the loudspeaker, has the sensation of clearly and loudly hearing the bass frequencies.

The third inventive embodiment, shown in Figure 3, will now be described and explained.

In the third embodiment, in Figure 3, the nonlinear circuit NL1 is shown in detail. The nonlinear circuit NL1 is composed of a nonlinear circuit NL2 and a function generator F2. The input signal of the nonlinear circuit NL1 - that is the output signal of the amplifier AM - is conducted to the input of the nonlinear circuit NL2 and the input of the function generator F2, whose output is connected to the control input of the nonlinear circuit NL2. The signal output of the nonlinear circuit NL2 is connected to the input of the bandpass filter BP2, while the control output of the nonlinear circuit NL2 is connected to the function generator F1 or to the low-pass filter TP1.

The nonlinear circuit NL2 continuously generates harmonics of the low-frequency components of the audio signal, which are weighted with the variable factor NG by the function generator F2. The factor NG is a function of the input signal. Depending on the choice of function for the function generator F2, manifold acoustic effects can be created.

For example, the function generator F2 can be designed so that the nonlinear circuit NL2 more strongly generates harmonics as soon as the signal amplitude must be limited, so as to prevent overdrive. In this way, the signal energy is distributed among the higher frequency harmonics, which a loudspeaker or a loudspeaker system can reproduce better. Although now the energy of the lower frequency signal components is reduced, the listener nevertheless has the impression of a full bass sound due to the above-mentioned psychoacoustic effects.

The fourth inventive embodiment, shown in Figure 4, will now be described and explained.

Figure 4 shows in detail an exemplary structure of the nonlinear circuit NL2 and an exemplary structure of the function generator F2.

The input signal to the nonlinear circuit NL1, the output signal from the amplifier AM, is conducted to the first input of an adder A2, to the input of an absolute value forming circuit ABS, and to the input of a peak value detector PK, whose output is connected to the input of a low-pass filter TP2 and a low-pass filter TP3. The output of the low-pass filter TP2 is connected to the first input of a subtractor S, and the output of the low-pass filter TP3 is connected to the second input of a subtractor S. The output of the subtractor S is connected, via a limiter LIM1, to the first input of a multiplier M2. The output of the absolute value forming circuit ABS is connected to the second input of the multiplier M2, whose output is connected to the second input of the adder A2. The output of the adder A2 is connected to the input of a limiter LIM2, whose control output delivers the control variable V to the function generator F1 or to the low-pass filter TP1, and whose output is connected to the input of the bandpass filter BP2. The audio signal for a loudspeaker or a loudspeaker system is available at the output of the bandpass filter BP2.

The peak value detector PK determines the level of the maximum amplitude occurring during a specified time interval T. The output signal of the peak value detector PK is time-averaged by the two low-pass filters TP2 and TP3. The time constant of the low-pass filter TP3 is smaller than that of the low-pass filter TP2, i.e. the cut-off frequency of the low-pass filter TP3 with the smaller time constant is higher than that of the low-pass filter TP2 with the larger time constant. Because of the smaller time constant, the output signal of the low-pass filter TP3 follows a change of the input signal faster than does the output signal of the low-pass filter TP2. The absolute value forming circuit ABS forms the absolute value of the input signal, which is weighted in the multiplier M2 by the factor NG, which has been generated by the subtractor S. A limiter LIM1 limits the factor NG to arrange between 0 and 1. The weighted absolute value of the input signal is added in the adder A2 to the input signal, and the resulting sum is limited to a specified amplitude by means of the limiter LIM2, so as to prevent overdrive.

For example, if the amplitude of the input signal rises discontinuously, the level at the output of the low-pass filter TP3 will rise faster, due to its smaller time constant, than at the output of the low-pass filter TP2. As a result, the factor NG, which is to be regarded as a control variable, assumes a positive value for rising amplitudes in the input signal. The stronger the amplitude of the input signal rises, the more harmonics will be generated and will be admixed to the input signal. On the other hand, if the amplitude falls, the factor NG becomes negative, because now the level at the output of the low-pass filter TP3, due to its smaller time constant, becomes smaller than the level at the output of the low-pass filter TP2. Because the factor NG has a lower limit of zero, no harmonics are admixed to the audio signal when the amplitudes are falling.

A significant advantage of the invention is that the nonlinear operation of the nonlinear circuit NL1, and the function of the function generator F1, determine the form of the harmonics as well as the time of their generation. By skillful choice of the nonlinear operation of the nonlinear circuit and of the function of the function generator, the invention can easily be adapted to loudspeakers with different characteristics, so that optimum compensation of the frequency response of a loudspeaker is always achieved. Because the amplitude of the audio signal is limited to a specified value by the nonlinear circuit NL1, the inventive circuit reacts much faster than the prior art to rising amplitudes of the audio signal.

The invention is especially suited for acoustic reproduction units, e.g. television units, portable radios, which are equipped with loudspeakers with a weak bass range, because the invention prevents overdriving the entire reproduction system and at the same time offers the listener the illusion of sonorous basses, even though the loudspeakers really do not radiate these low bass frequencies.

List of Reference Symbols

P1	First path
P2	Second path
BP1	First bandpass filter
BP2	Second bandpass filter
M1	First multiplier
M2	Second multiplier
AM	Amplifier
NL1	First nonlinear circuit
NL2	Second nonlinear circuit
F1	First function generator
F2	Second function generator
G	Amplification factor
GC	Correction factor
NG	Factor
V1, V2	Control variable
ABS	Absolute value forming circuit
PK	Peak value detector
TP1	Low-pass filter
TP2	Low-pass filter
TP3	Low-pass filter
S	Subtractor
LIM1	Limiter
LIM2	Limiter
Xin	Audio signal
Xout	Output signal

Claims

1. A method for processing an audio signal (Xin), which, via a first path (P1), is conducted to the first input of a first adder (A1), characterized in that the audio signal (Xin) is band-limited in a second path (P2) by means of a first bandpass filter (BP1) , and that, at the output of the first bandpass filter (BP1), the band-limited audio signal is multiplied by a correction factor (GC), is amplified by an amplification factor (G), and then its amplitude is limited to a specified maximum by means of a nonlinear circuit (NL1), and that the correction factor (GC) is reduced when the prescribed maximum is exceeded, but otherwise remains constant or is increased, and that the audio signal at the output of the first nonlinear circuit (NL1) is band-limited by means of a second bandpass filter (BP2), and that the band-limited audio signal at the output of the second bandpass filter (BP2) is added, in the adder (AD), to the audio signal of the first path (P1).

2. The method of Claim 1, characterized in that the correction factor (GC) is a function (F1) of a control variable (V), which is generated by the first nonlinear circuit (NL1).

3. The method of Claim 2, characterized in that the correction factor (GC) assumes a first value (V1) if the amplitude is less than the prescribed threshold, and assumes a second value (V2) if the amplitude is greater than the specified threshold.

4. The method of Claim 2 or 3, characterized in that the function (F1) is implemented by means of a first low-pass filter (TP1).

5. The method of one of the Claims 1 to 4, characterized in that the nonlinear circuit (NL1) generates harmonics of the low-frequency signal components and weights them with a variable factor (NG).

6. The method of Claim 5, characterized in that the variable factor (NG) is a function (F2) of the input signal of the first nonlinear circuit (NL1).

7. The method of Claim 6, characterized in that the first nonlinear circuit (NL1) increasingly generates harmonics at the beginning of a low-frequency signal.

8. The method of Claim 7, characterized in that the input signal of the first nonlinear circuit (NL1) is detected by a peak value detector (PK) in the first nonlinear circuit (NL1), in accordance with the function (F2) of the input signal of the first nonlinear circuit (NL1), and that the output signal of said peak value detector is low-pass filtered in a second and third low-pass filter (TP2, TP3) with different time constants, and that the difference signal of the two low-pass filtered signals represents the variable factor (NG), and that the absolute value of the input signal of the first nonlinear circuit (NL1) is weighted with this factor (NG) and is added to the input signal of the first nonlinear circuit (NL1) to form a summation value, whose amplitude is limited to a specified value.

9. A circuit arrangement to process an audio signal (Xin) which, via a first path (P1), is conducted to the first input of a first adder (A1), whose output delivers the output signal (Xout), characterized in that the audio signal (Xin) is conducted to the second input of the first adder (A1) via a second path (P2), consisting of a first bandpass filter (BP1), a first multiplier (M1), an amplifier (AM) with variable amplification, a first nonlinear circuit (NL1), and a second bandpass filter (BP2), all connected in series, and that a control output of the first nonlinear circuit (NL1) is connected to the input of a first function generator (F1), whose output is connected to the first multiplier (M1), and that an amplification (G) is applied to the control input of the amplifier (AM).

10. The circuit arrangement of Claim 9, characterized in that the first function generator (F1) is designed as a first low-pass filter (TP1).

11. The circuit arrangement of Claim 9 or 10, characterized in that the first nonlinear circuit (NL1) consists of a second nonlinear circuit (NL2) and the second function generator (F2), and that the input signal of the first nonlinear circuit (NL1) is applied to the input of the second nonlinear circuit (NL2) and to the input of the second function generator (F2), whose

output is connected to the control input of the second nonlinear circuit (NL2), and that the control output of the second nonlinear circuit (NL2) is connected to the first function generator (F1) or to the first low-pass filter (TP1), and that the signal output of the second nonlinear circuit (NL2) is connected to the input of the second bandpass filter (BP2).

12. The circuit arrangement of Claim 11, characterized in that the second function generator (F2) consists of a peak value detector (PK), a second and a third low-pass filter (TP2, TP3), a first limiter (LIM1), and a subtractor (S), and that the second nonlinear circuit (NL2) consists of an absolute value forming circuit (ABS), a second multiplier (M2), a second adder (A2), and a second limiter (LIM2), and that the input signal of the first nonlinear circuit (NL1) is conducted to the first input of the second adder (A2), to the input of the absolute value forming circuit (ABS), and to the input of the peak value detector (PK), whose output is connected to the input of the second low-pass filter (TP2) and to the input of the third low-pass filter (TP3), and that the output of the second low-pass filter (TP2) is connected to the first input of a subtractor (S), and the output of the third low-pass filter (TP3) is connected to the second input of a subtractors (S), whose output is connected, via the first limiter (LIM1), to the first input of the second multiplier (M2), and that the output of the absolute value forming circuit (ABS) is connected to the second input of the second multiplier (M2), whose output is connected to the second input of the second adder (A2), and that the output of the second adder (A2) is connected to the input of the second limiter (LIM2), whose control output is connected to the first function generator (F1) or to the first low-pass filter (TP1), and whose output is connected to the input of the second bandpass filter (BP2).

Abstract of the Disclosure

So as to compensate the frequency response of loudspeakers and so as to give a listener the illusion of sonorous bass tones, it is known that an audio signal can be divided into a first and second path, such that harmonics of the signal components with a low-frequency are generated in the second path and are admixed to the signal in the first path. To improve reproduction, especially in the bass range of weakly designed loudspeakers, the audio signal is bandpass filtered (BP1), weighted with a correction factor GC (M1), amplified with an amplification factor G (AM), then limited to a maximum value (NL1), and finally again bandpass filtered (BP2), in the second path (P2), before it is added to the original audio signal (Xin) in the first path (P1). The correction factor (GC) is reduced when the maximum value is exceeded, while otherwise it remains constant or is increased. Through this measure, harmonics are generated at the onset of a low-frequency audio signal, and are admixed to the original audio signal.

Figure 1

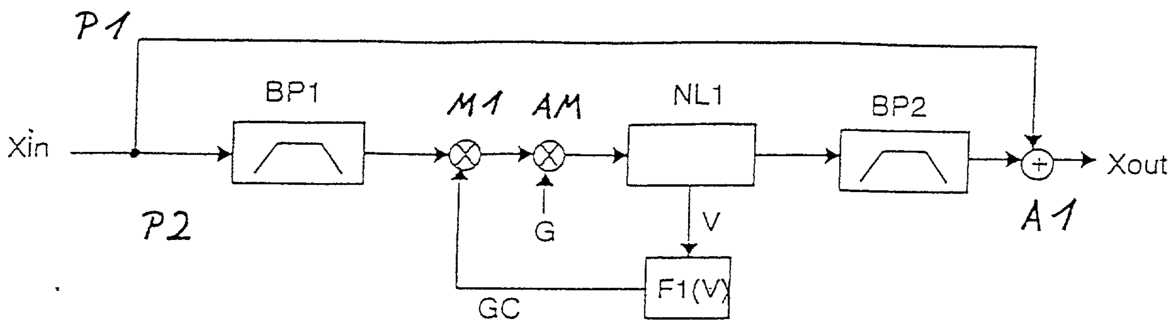
P1
BP1
M1
AM
NL1
BP2
Xin
Xout
P2
G
V
A1
F1(V)
GC

Fig. 1

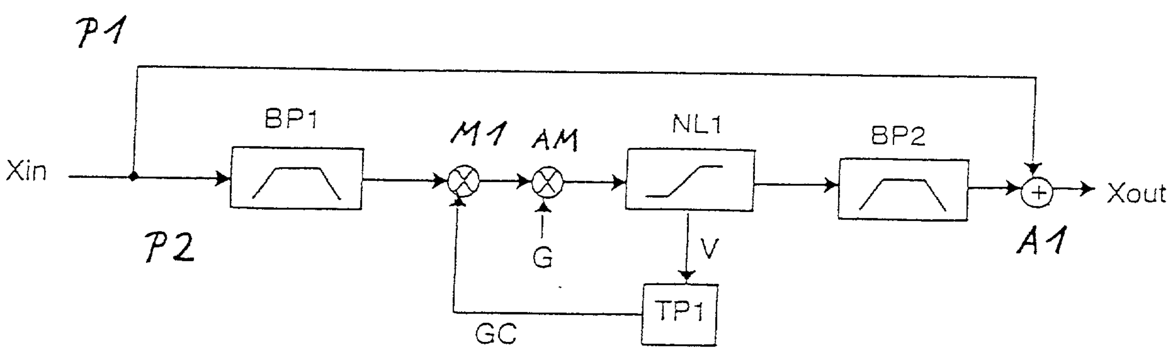
P1
BP1
M1
AM
NL1
BP2
Xin
Xout
P2
G
V
A1
TP1
GC

Fig. 2

10/030521

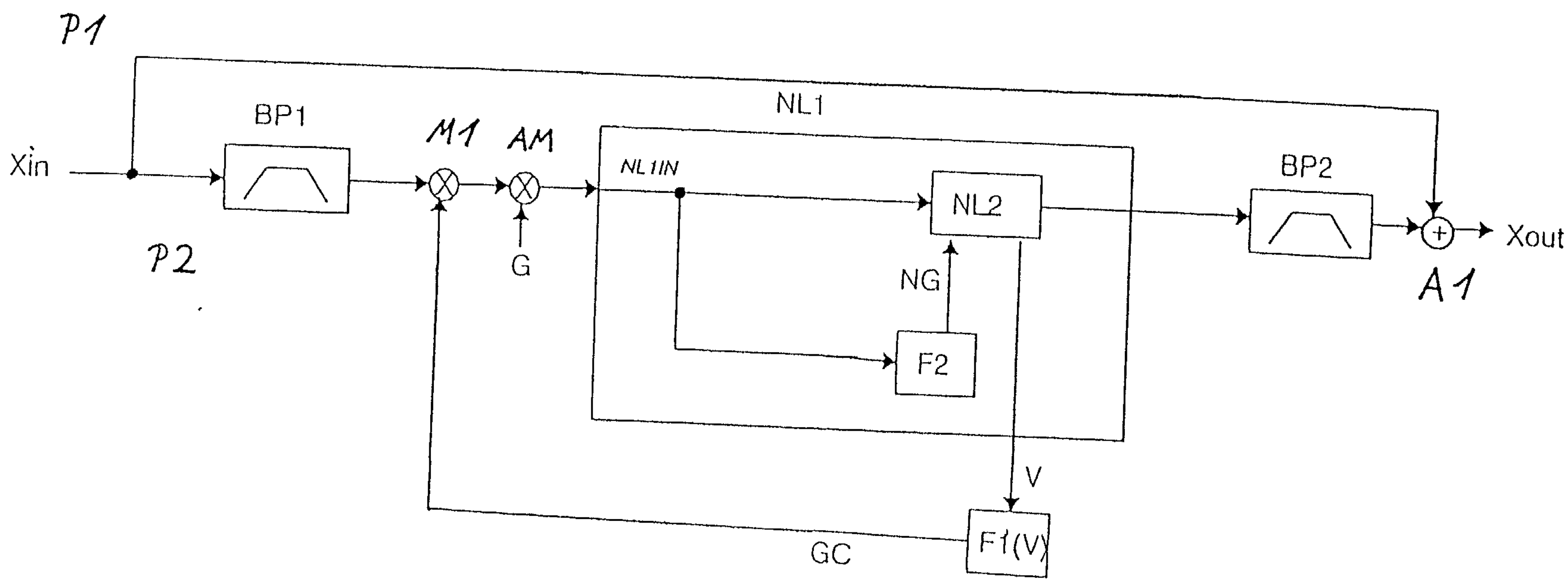

Fig. 3

10/030521

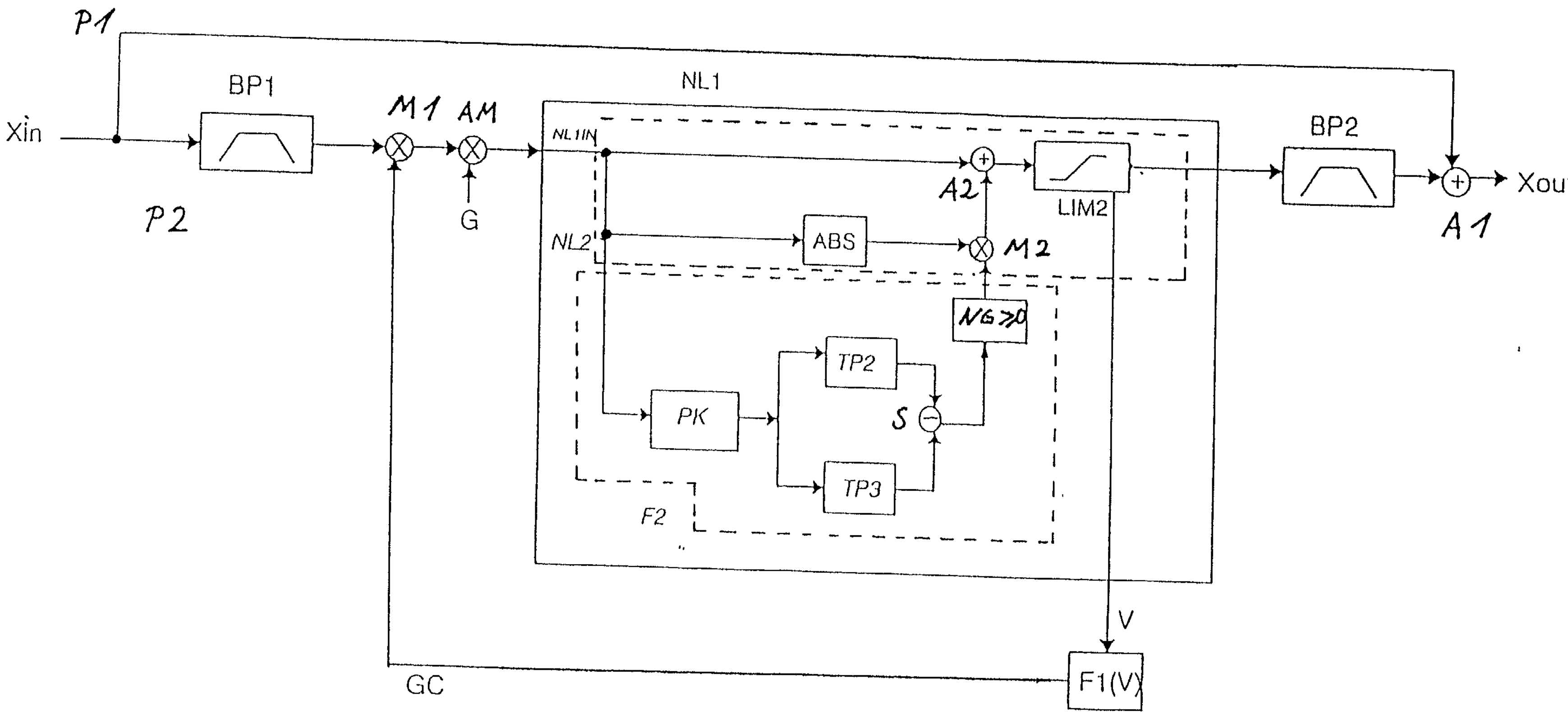

Fig. 4

Micronas.6055

DECLARATION AND POWER OF ATTORNEY

We, the below named inventors, hereby declare that:

Our residences, post office addresses, and citizenships are as stated below next to our respective names.

We believe we are the original, first, and joint inventors of the subject matter which is claimed and for which a patent is sought on the invention entitled **A METHOD FOR PROCESSING AN AUDIO SIGNAL**, the specification of which was filed with the United States Patent and Trademark Office on December 26, 2001 and designated Serial No. 10/030,521.

We hereby state that we have reviewed and understand the contents of the above-identified specification, including the claims.

We acknowledge the duty to disclose information which is material to patentability in accordance with Title 37, Code of Federal Regulations, Section 1.56.

We hereby claim foreign priority benefits under Title 35, United States Code §119(a)-(d) or §365(b) of any foreign application(s) for patent or inventor's certificate listed below and have also identified below any foreign application for patent or inventor's certificate filed by us on the same subject matter having a filing date before that of the application on which priority is claimed: International Patent Application No. PCT/EP 00/04798 filed May 26, 2000.

We hereby declare that all statements are made hereby of our own knowledge are true and that all statements made on information and belief are believed to be true; and further that these statements were made with the knowledge that willful false statements and the like so made are punishable by fine or imprisonment, or both, under Section 1001 of Title 18 of the United States Code and that such willful false statements may jeopardize the validity of the application or any patent issued thereon.

And we hereby appoint:

Maurice E. Gauthier	-	Reg. No. 20,798
Richard L. Stevens	-	Reg. No. 24,445
Matthew E. Connors	-	Reg. No. 33,298
William E. Hilton	-	Reg. No. 35,192
Patrick J. O'Shea	-	Reg. No. 35,305
Arlene J. Powers	-	Reg. No. 35,985
Richard J. Stevens, Jr.	-	Reg. No. 44,357
Peter Stecher	-	Reg. No. 47,259

all of the firm of Samuels, Gauthier & Stevens, our attorneys with full power of substitution and revocation, to prosecute this application and to transact all business in the Patent and Trademark Office connected therewith.

We request that all correspondence be directed to:

Patrick J. O'Shea, Esq.
Samuels, Gauthier & Stevens, LLP
225 Franklin Street, Suite 3300
Boston, MA 02110

1-0 Matthias Vierthaler — April 09, 2002
Matthias Vierthaler — Date

Residence:	Blasiusstrasse 2 D-79108 Freiburg GERMANY DEX
Citizenship:	German
Post Office Address:	Same as above

2-0 Martin Winterer — April 10, 2002
Martin Winterer — Date

Residence:	Bundesstrasse 106 D-79194 Gundelfingen GERMANY DEX
Citizenship:	German
Post Office Address:	Same as above